VoIP Analog Telephone Adapter

VIP-156/VIP156PE/VIP-157/VIP-157S

User's manual

Version 3.31

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CE mark Warning

The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

WEEE Warning

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Revision

User's Manual for PLANET VoIP Analog Telephone Adapter:

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TABLE OF CONTENTS

Chapter 1 Introduction 6

Overview 6

Package Content 7

Physical Details 7

LED Display & Button 10

Chapter 2 Preparations & Installation 12

Physical Installation Requirement 12

LAN IP address configuration via web configuration interface 13

Chapter 3 Network Service Configurations 17

Configuring and monitoring your ATA from web browser 17

Overview on the web interface of ATA 17

Manipulation of ATA via web browser 17

Chapter 4 VoIP Telephone Adapter Configurations 19

Phone Book 19

Call Forward 20

SNTP settings 22

Volume Setting 22

Block Setting 23

Auto Answer (For VIP-157) 24

Caller ID settings 24

Dial Plan Settings 25

Flash Time Setting 28

Call waiting Settings 28

T.38 (FAX) Settings 29

Hot line Settings 29

Alarm Settings 30

LAN Settings 30

PC Settings 31

DDNS Settings 32

VLAN Settings 33

DMZ Settings 34

Virtual Server 34

PPTP Settings 35

Service Domain Settings 36

Port Settings 37

Codec Settings 37

Codec ID Setting 38
DTMF Setting 39
RPort Settings 39
Other Settings 40
STUN settings 40
Auto Configuration 41
Encrypt Config 42
PTT Settings 42
Cancel without to tag 43
MAC CIone Setting 43
Tone Settings 45
Advanced Settings 45
System Authority 46
Save & Reboot 46
Firmware Upgrade 47
Auto Upgrade 47
Reset to Default 49
Reboot without saving 49
Appendix A Voice Communication Samples 51
Case 1: ATA to ATA connection via IP address 51
Case 2: (Peer-to-Peer mode) VIP-157S Port 1 to Port 2 communications 52
Case 3: Voice communication via SIP proxy server – SIP-50 53
Case 4: Voice communication via IP PBX system – IPX-2000 (Auto-config) 55
Case 5: Call Forward Feature_Example 1 59
Case 6: Call Forward Feature_Example 2 61
Case 7: Call Forward Feature_Example 3 63
Case 8: Call Forward Feature_Example 4 65
Case 9: Auto Answer Feature_IP to PSTN 66
Case 10: Auto Answer Feature_PSTN to IP 68
Appendix B The method of operation guide 70
Call Transfer 70
3-Way Conference 70
Call Waiting 70
Switch the Realm (Registration Proxy Server) 70
Auto Update firmware by manual (Keypad) 71
Appendix C VIP-156/VIP-156PE/VIP-157/VIP-157S Specifications 72

Chapter 1 Introduction

Overview

Based on years of VoIP manufacturing experiences, PLANET Technology VoIP total solutions are known for advanced implementation of standards based telephony with mass deployment capability.

Cost-effective, easy-to-install and simple-to-use, the PLANET VIP-156/VIP-157/VIP-157S VoIP Phone Adapter ("**ATA**" in the following term) converts standard telephones to IP-based networks. The service providers and enterprises offer users traditional and enhanced the telephony communication services via the existing broadband connection to the Internet or corporation network.

With the ATA, home users and companies are able to save the installation cost and extend their past investments in telephones, conference and speakerphones. The ATA can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

The ATA includes two alternatively Ethernet interface for Internet (PPPoE, DHCP or Fixed IP), or office LAN connection. With adding the auto-provision feature of our IP PBX product - IPX-2000, the ATA can be seamlessly integrated into the telephony network and be used in consumer and business IP telephony service, no expertise required!

The ATA and our IP PBX system integration are the ideal combination for your office daily communications.

Product Features

- Feature-rich telephone service over home or office Internet/Intranet connection
- Auto-config feature for flexible, ease-of-use IP PBX system integration
- Cost effective, field proven compatibility, and stability
- Web-based and telephone keypad machine configuration
- Remote machine administration authentication
- Voice prompt for machine configurations

VoIP Features

- SIP 2.0 (RFC3261) compliant
- Peer-to-Peer / SIP proxy calls
- Voice codec support: G.711, G.723.1, G.729A/G.729B
- T.38 FAX transmission over IP network

- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation (CNG)
- In band, out-of-band, and SIP-info DTMF support

Package Content

The contents of your product should contain the following items:
VoIP Telephone Adapter
Power adapter
Quick Installation Guide
User's Manual CD
RJ-11 cable x 1

Physical Details

The following figure illustrates the front/rear panel of ATA.

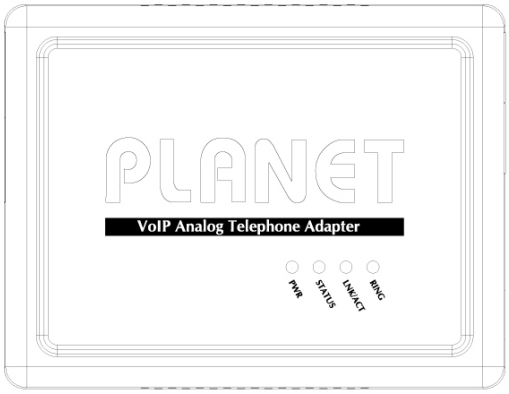
Respective model/descriptions are shown below:

VIP-156: SIP Analog Telephone Adapter

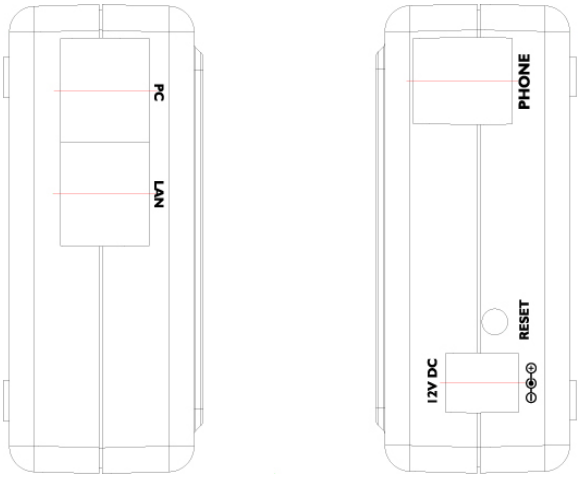
VIP-156PE: 802.3af PoE SIP Analog Telephone Adapter

VIP-157: 1 FXS/ 1 FXO SIP Analog Telephone Adapter

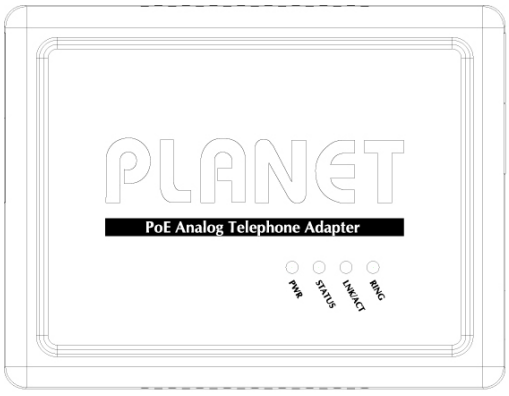
VIP-157S: 2-port FXS SIP Analog Telephone Adapter



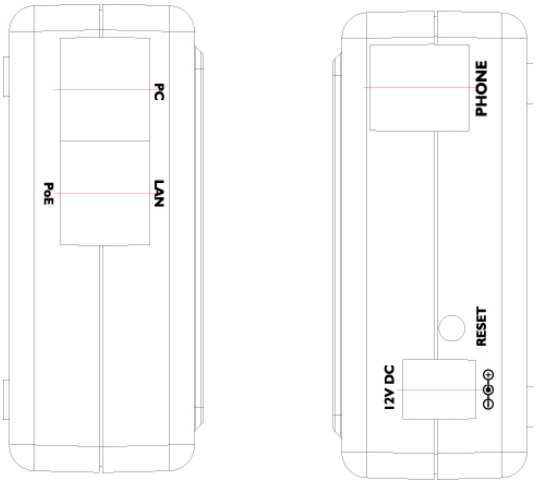
Front Panel of VIP-156



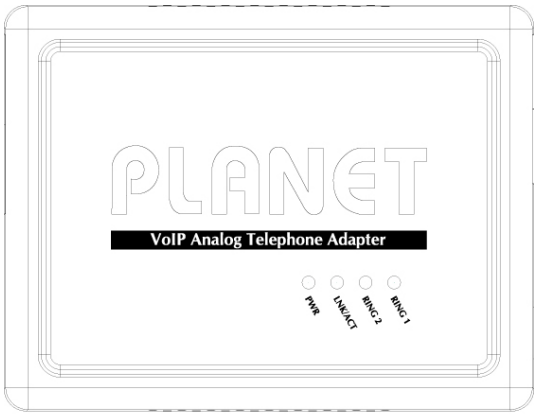
Left / Right Panel of VIP-156



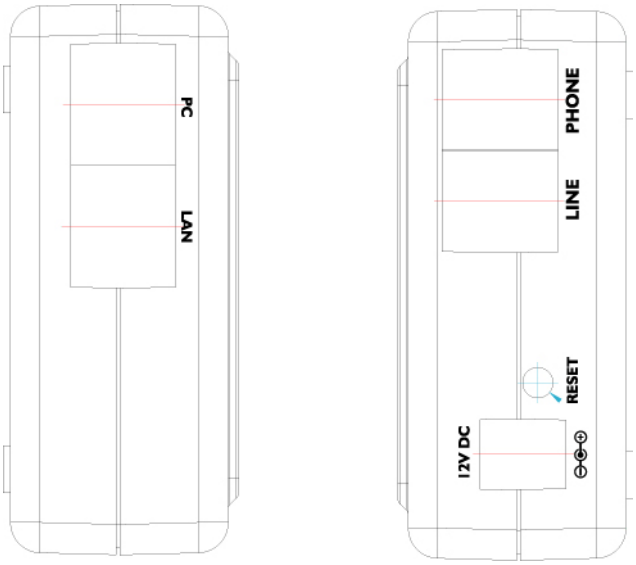
Front Panel of VIP-156PE



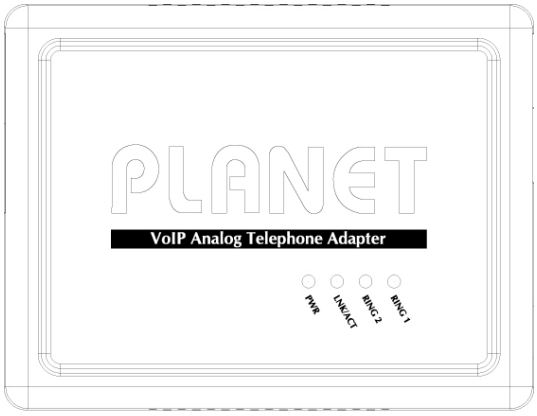
Left / Right Panel of VIP-156PE



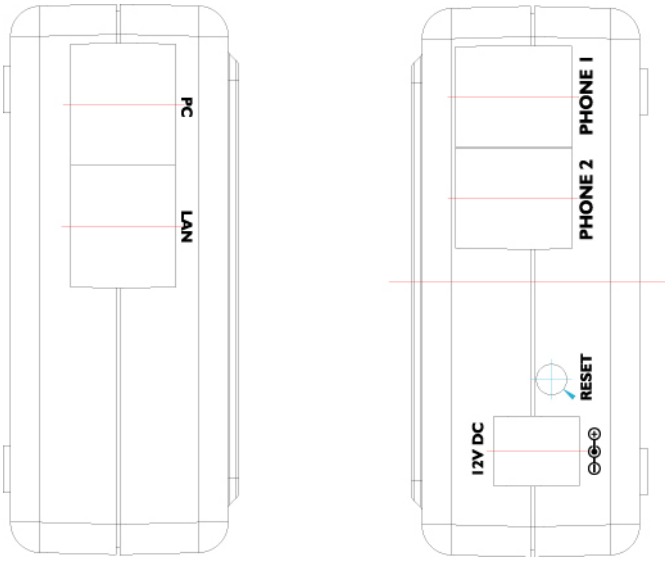
Front Panel of VIP-157



Left / Right Panel of VIP-157



Front Panel of VIP-157S



Left / Right Panel of VIP-157S

LED Display & Button

1	**PC**	RJ-45 connector, to maintain the existing network structure, connected directly to the **PC** through **straight** CAT-5 cable
2	**LAN**	RJ-45 connector, for Internet access, connected directly to **Switch/Hub** through **straight** CAT-5 cable. The **LAN** interface also can be connected with 802.3af PoE switch or converter for power supply (**VIP-156PE**)
3	**12V DC**	12V DC Power input outlet
4	**Reset**	Reset to the factory default setting

Note

Machine default IP is http://192.168.0.1. Press **RESET** button on rear panel over 5 seconds will reset the VoIP Phone Adapter to factory default value. (Except speed dial and call forward settings)

LED display of VIP-156 / VIP-156PE

LED Indicators	Descriptions
PWR	Power is supplied to the device.
STATUS	The Status LED will be flashing when the machine is operational
LNK/ACT	**OFF**: the device is connected to LAN at 10Mb/s. **ON**: the device is connected to LAN at 100Mb/s.
RING	**OFF**: the phone is idle. **ON**: the phone is in use (offhook). **Blinking**: the phone is ringing.

LED display of VIP-157 / VIP-157S

LED Indicators	Descriptions
STATUS	The Status LED will be flashing when the machine is operational
LNK/ACT	**OFF**: the device is connected to LAN at 10Mb/s. **ON**: the device is connected to LAN at 100Mb/s.
RING 1	**OFF**: the phone is idle. **ON**: the phone is in use (offhook). **Blinking**: the phone is ringing.
RING 2	**OFF**: the phone is idle. **ON**: the phone is in use (offhook). **Blinking**: the phone is ringing.

Chapter 2
Preparations & Installation

Physical Installation Requirement

This chapter illustrates basic installation of ATA analog Phone Adapter (("**ATA**" in the following term))

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem

Administration Interface

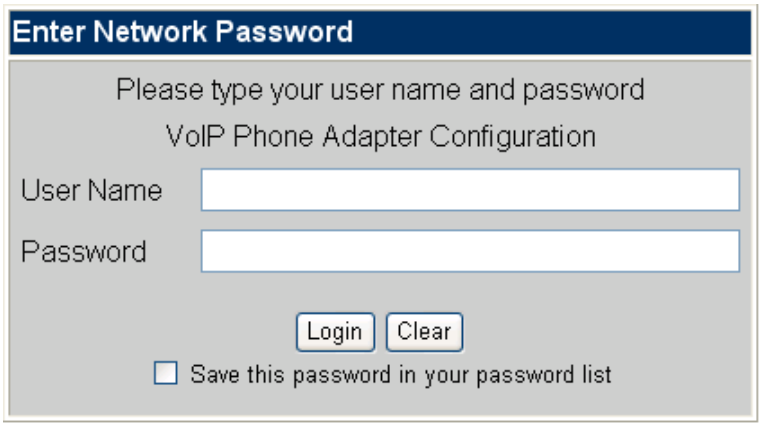
PLANET ATA provides GUI (Web based, Graphical User Interface) for machine management and administration.

Web configuration access

To start ATA web configuration, you must have one of these web browsers installed on computer for management

- Microsoft Internet Explorer 6.0.0 or higher with Java support

Default LAN interface IP address of ATA is **192.168.0.1**. You may now open your web browser, and insert http://192.168.0.1 in the address bar of web browser to logon ATA web configuration page.



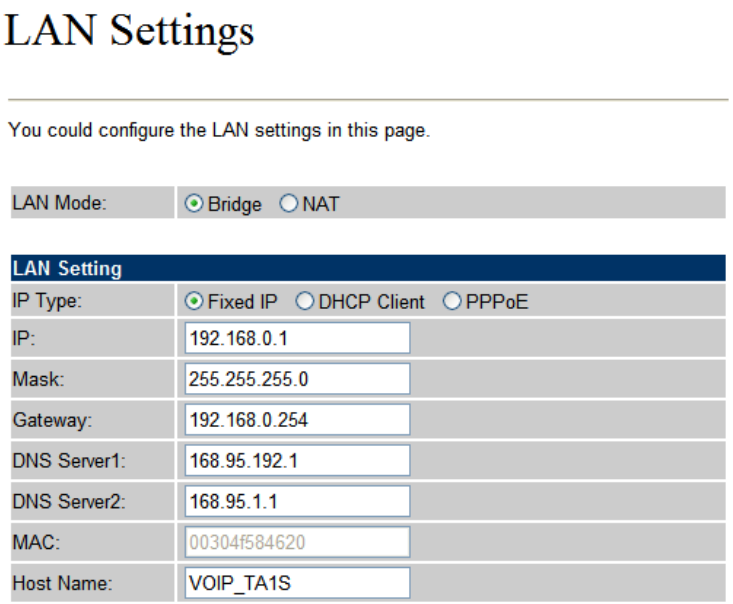
ATA will prompt for logon username/password, please enter: **root** / **null (no password)** to continue machine administration.

Note

Please locate your PC in the same network segment (**192.168.0.x**) of ATA. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of VIP in the adddress bar. After logging on machine with username/password (default**: root / no passwrd**), browse to "**Network**" --> "**LAN Settings**" configuration menu:



Parameter Description

IP address LAN IP address of ATA
Default: 192.168.0.1

Subnet Mask LAN mask of ATA
Default: 255.255.255.0

Default Gateway Gateway of ATA
Default: 192.168.0.254

Network settings via Keypad commands

The ATA series phone adapters support telephone keypad configurations, please connect analog telephone set and refer to the following table for machine network configurations.

IVR Menu Choice	Machine operation	Parameter(s)	Notes:
#111#	Set DHCP client	None	ATA will change to DHCP

			Client
#112xxx*xxx*xxx*xxx#	Setup Static IP Address	Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
#113xxx*xxx*xxx*xxx#	Set Network Mask	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#114xxx*xxx*xxx*xxx#	Set Gateway IP Address	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#115xxx*xxx*xxx*xxx#	Set Primary DNS Server	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#190#	Unlock	None	Must unlock the protect function before set up network settings and ATA function via keypad.
#191#	Lock	None	The system will be lock and can't set up network settings via keypad.
#195#	Reboot	None	The system will reboot automatically.
#198#	Factory Reset	None	The system will be reset to factory default value and reboot automatically.
0＊	To switch PSTN mode	None	VIP-157 only

Following keypad commands can be used to display the network settings enabled on ATA via voice prompt.

IVR Menu Choice	Machine operation	Notes:
#120#	Check PC IP Address	IVR will announce the current PC-port IP address of the ATA.
#121#	Check network connection Type	IVR will announce if DHCP in enabled or disabled.
#122#	Check the Phone Number	IVR will announce current enabled VoIP number.
#123#	Check Network Mask	IVR will announce the current network mask of the ATA.
#124#	Check Gateway IP Address	IVR will announce the current gateway IP

		address of the ATA.
#125#	Check Primary DNS Server Setting	IVR will announce the current setting in the Primary DNS field.
#126#	Check LAN IP Address	IVR will announce the current LAN port IP address of the ATA.
#128#	Check Firmware Version	IVR will announce the version of the firmware running on the ATA.

Following keypad commands can be used to set up the main function .

IVR Menu Choice	Machine operation	Parameter(s)	Notes:
#138#	Enable call waiting	None	Enable Call waiting
#139#	Disable call waiting	None	Disable Call waiting
#160#	Update firmware	None	Update firmware
#510#	Blind Transfer	ATA Bland Transfer	
#511#	Attendant Transfer	ATA Attendant Transfer	
#512#	3-way calling	ATA 3-way calling	
#514#	IP transfer to PSTN	ATA transfer IP call to PSTN side	
#130+[1~8]#	Set Codec	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
#131+[00~15]#	Set Handset Gain	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 10
#132+[00~12]#	Set Handset Volume	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10
#135xxx*xxx*xxx*xxx#	TFTP Server IP Address	Set Auto config TFTP Server IP Address	You can set the TFTP Server IP address
#136xxx*xxx*xxx*xxx#	FTP Server IP Address	Set Auto config FTP Server IP Address	You can set the FTP Server IP address
#137+[0~2]#	Auto config mode	0: Disable, 1: TFTP mode, 2: FTP mode	You can set the Auto configuration mode, 0: Disable, 1: use TFTP Server, 2: user FTP Server
#145#	Forward function disable	Disable forwrad funciton	
#146+Number#	enable forward to FXS	Eanble forward to FXS	

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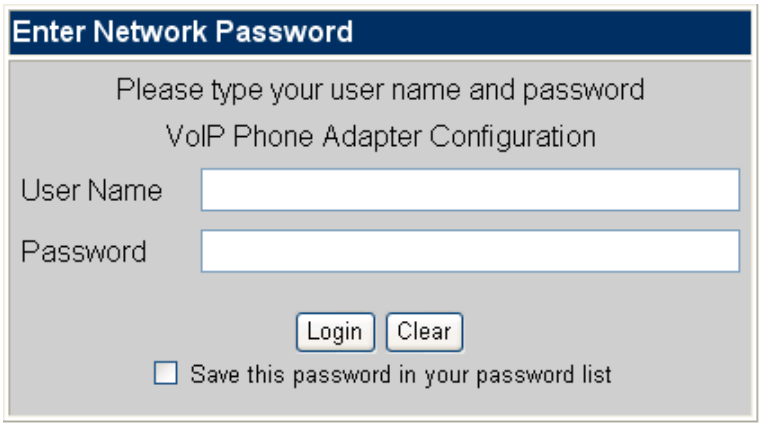


ATA will prompt for logon username/password, please enter: **root** / **null (no password)** to continue machine administration.

Chapter 3

Network Service Configurations

Configuring and monitoring your ATA from web browser

The ATA integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard web browser, you can configure and check machine status from anywhere around the world.

Overview on the web interface of ATA

With web graphical user interface, you may have:

- More comprehensive setting feels than traditional command line interface.
- Provides user input data fields, check boxes, and for changing machine configuration settings
- Displays machine running configuration

To start ATA web configuration, you must have one of these web browsers installed on computer for management

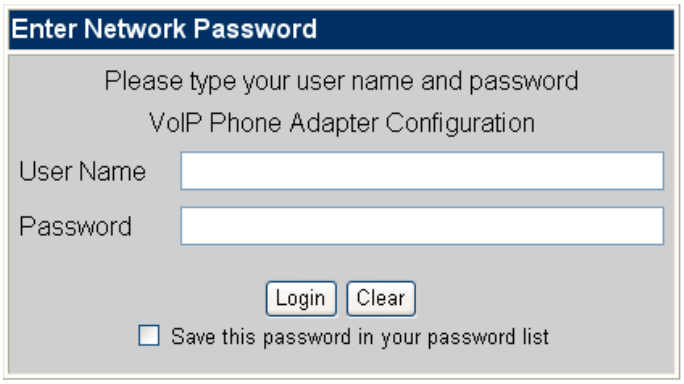
- Microsoft Internet Explorer 6.0.0 or higher with Java support

Manipulation of ATA via web browser

Log on ATA via web browser

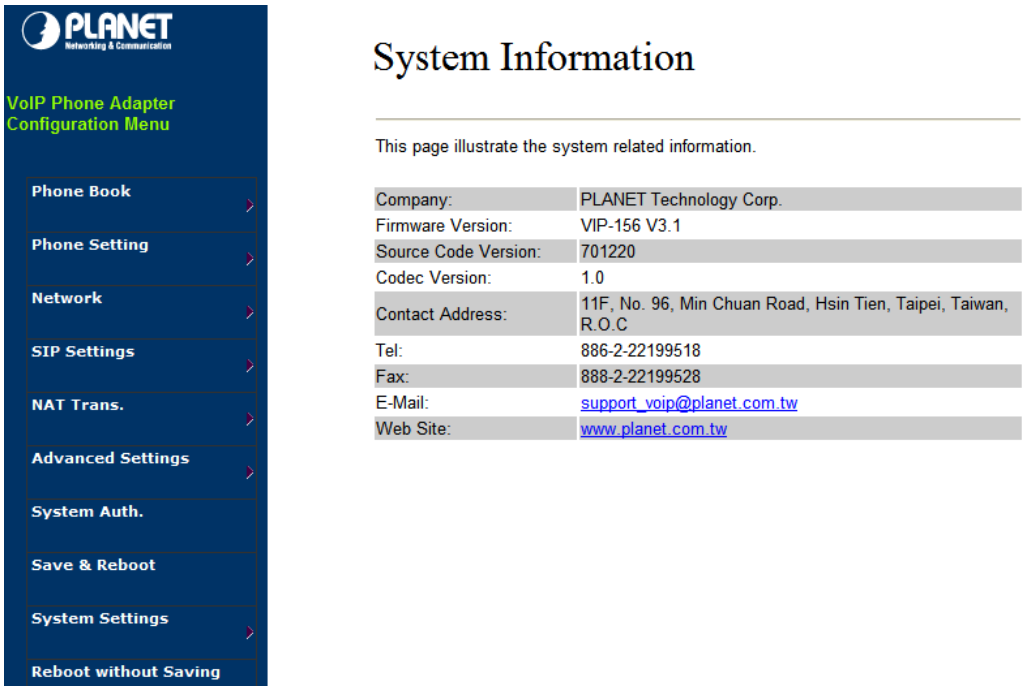
After TCP/IP configurations on your PC, you may now open your web browser, and input http://192.168.0.1 to logon Phone Adapter web configuration page.

Phone Adapter will prompt for logon username/password: ***root*** / ***null (no password)***



ATA log in page

When users login the web page, users can see the Phone Adapter system information like firmware version, company…etc in this main page.



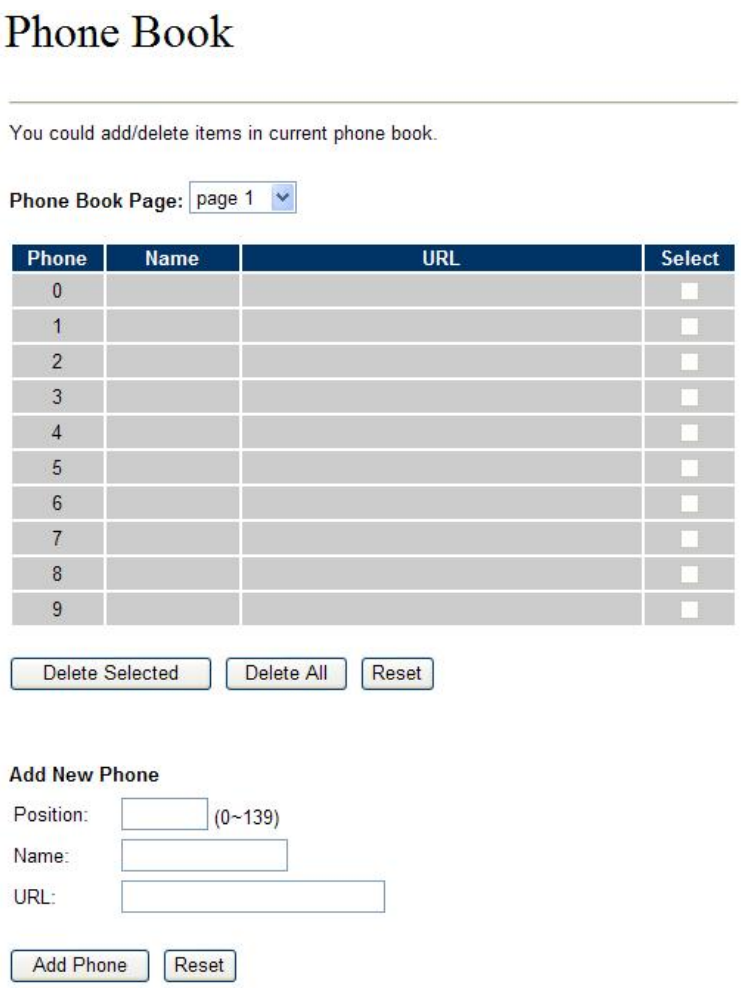
VoIP Phone Adatper main page

Chapter 4

VoIP Telephone Adapter Configurations

Phone Book

ATA can set up 140 records of Phone Book. User can dial the **Name** records to make calls via **Phone Book** feature.



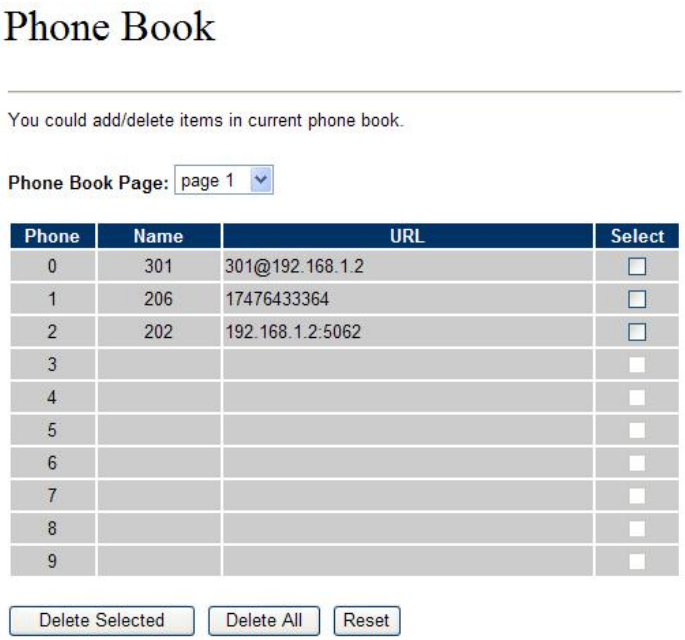
Field	Description
Phone Book Page	The default is Page 1. It can select Page1 ~ Page 14 to look round Phone Book records.
Phone	The record number from 0 ~ 139, it can set up 140 records in total.
Name	The name of Phone Book records, it only can input numerals.
URL	Fill in the outgoing number (Line Number) or IP address.
Select	To select this record.

If you need to add a phone number into the Phone Book list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.

If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

If you want to delete all phone numbers, you can click “Delete All” button.

For Example:



Ex_1:

ATA had added the above phone numbers. User pick up the handset and dial the “**301**” to make the P2P call (301@192.168.1.2).

Ex_2:

Users pick up the handset and dial the “**206**” to make the Proxy call (17476433364).

Ex_3:

Users pick up the handset and dial the “**202**” to make the P2P call (192.168.1.2:5062).

Call Forward

This page defines Call Forward function. You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

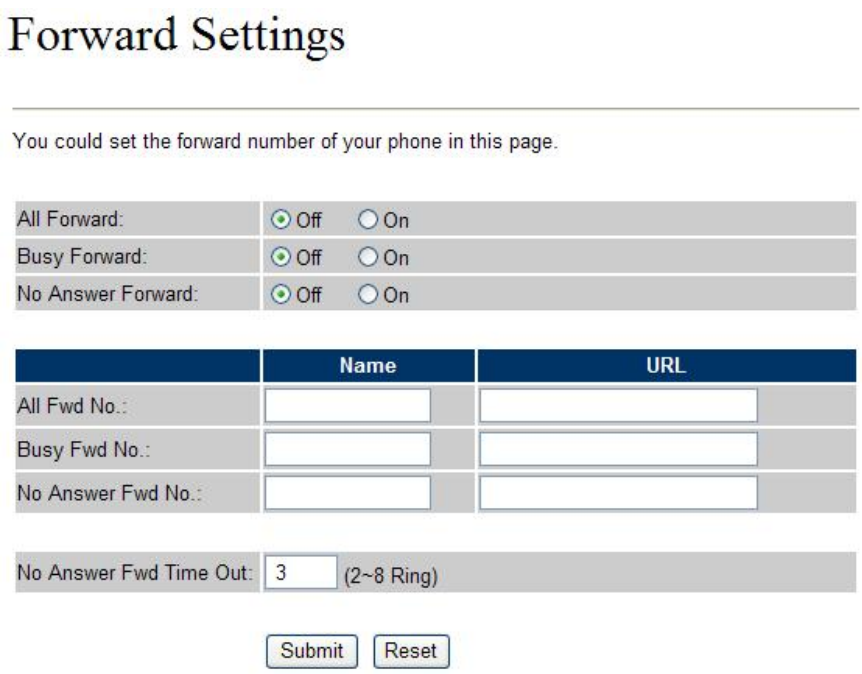
All Forward: All incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the

speed dial number you choose.

Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

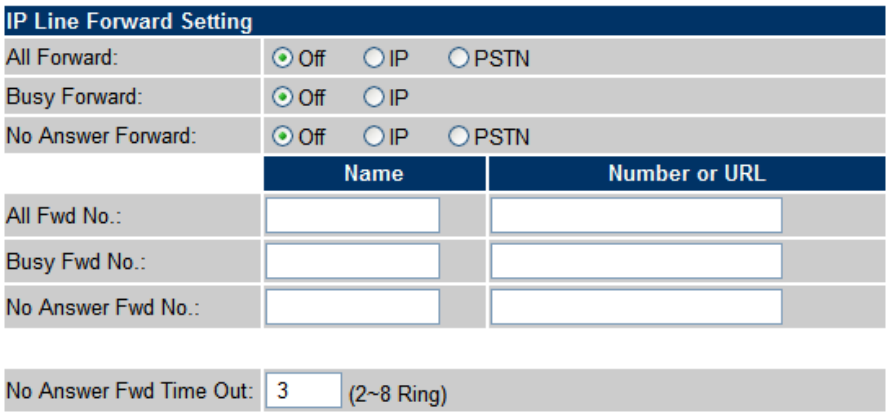
No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you chosen. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

When you finished the setting, please click the Submit button.



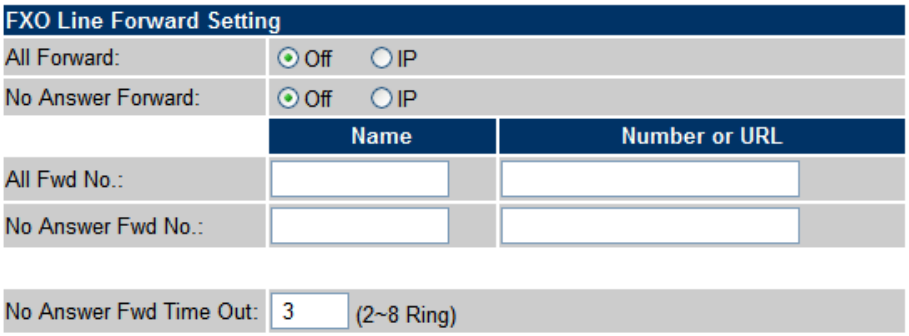
Call Forward function for VIP-156/VIP-156PT/VIP-157S

Call Forward to PSTN (VIP-157): VIP-157 not only support Call Forward to IP calls, but also can forward the calls to PSTN. You can choose the Call Forward type with PSTN, then input the name and the PSTN number in URL/Number field.


IP Line Forward Setting
All Forward: Off IP PSTN
Busy Forward: Off IP
No Answer Forward: Off IP PSTN
Name
Number or URL
All Fwd No.:
Busy Fwd No.:
No Answer Fwd No.:
No Answer Fwd Time Out: 3 (2~8 Ring)

IP Line Forward function for VIP-157

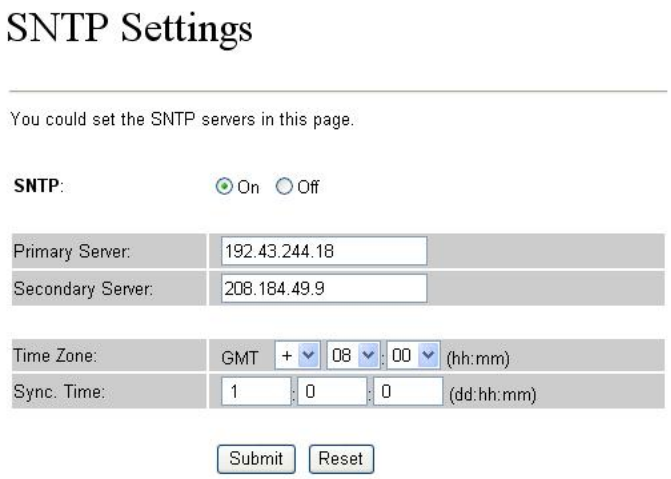
The IP Line Forward function is use for the incoming call is IP call type, and the destination is IP or PSTN call types. The FXO Line Forward function is use for the incoming call is PSTN call type, and the destination is IP call type. The IP / FXO Line Forward functions can be functioned at the same time, and that could separate different incoming call types for flexable applications.



FXO Line Forward function for VIP-157

SNTP settings

This page defines the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

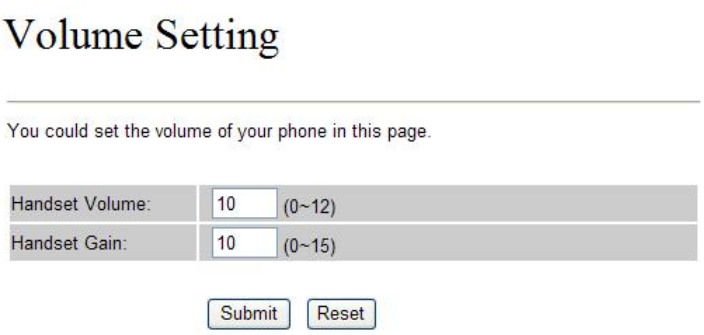


Volume Setting

This page defines the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

Handset Volume is to set the volume for you can hear from the handset.

Handset Gain is to set the volume send out to the other side's handset.

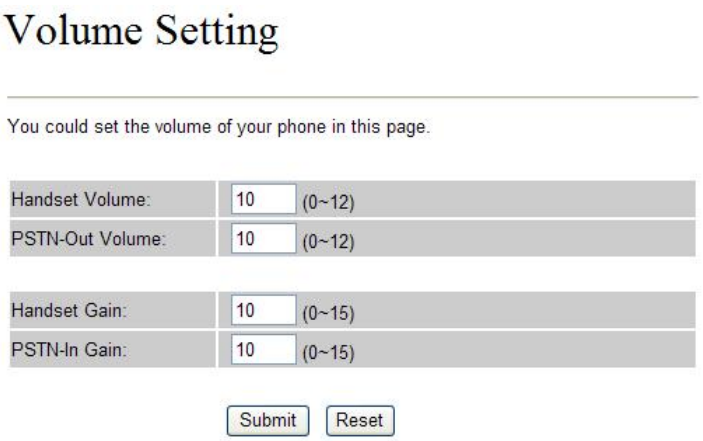


Volume Settings for VIP-154T/VIP-154PT

Beside the above settings, VIP-157 also can set the volume of PSTN.

PSTN-Out Volume is to set the PSTN volume for you can hear.

PSTN-In Gain is to set the volume send out to the other side's handset.



Volume Settings for VIP-157

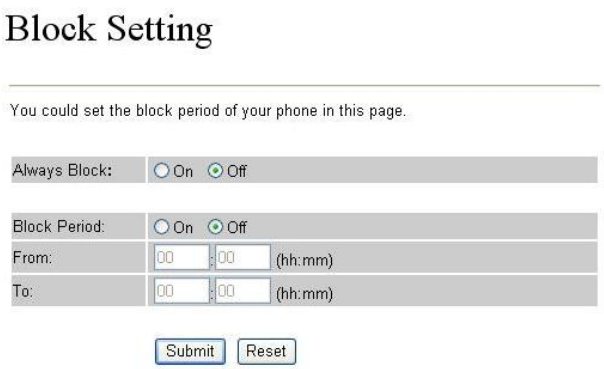
Block Setting

This page defines the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

Always Block: All incoming call will be blocked until disable this feature.

Block Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

When you finished the setting, please click the Submit button.


Block Setting
You could set the block period of your phone in this page.
Always Block: On Off
Block Period: On Off
From: 00 : 00 (hh:mm)
To: 00 : 00 (hh:mm)
Submit Reset

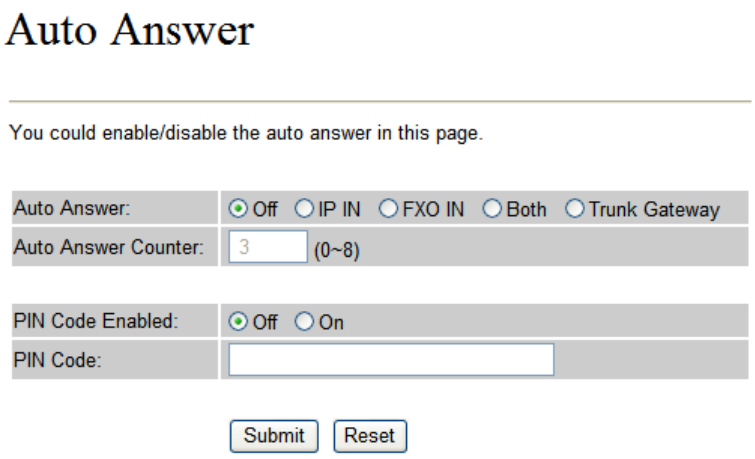
Auto Answer (For VIP-157)

This page defines the Auto Answer function. You can set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the VIP-157 can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the VIP-157 can let user to redial to IP Phone number.

Auto Answer: There are different incoming call types for flexable applications. The **Trunk Gateway** function needs to arrange in with the registered Server System. The 3-Party subscribers could make Off-Net call (PSTN) through the FXO port of VIP-157.

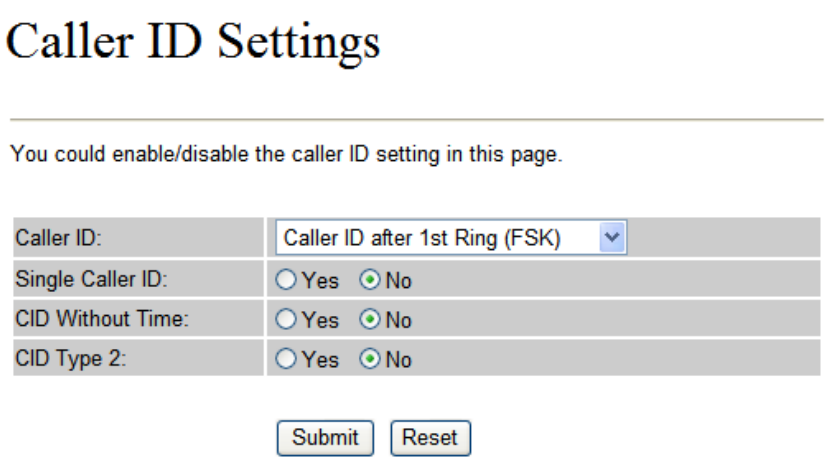
Auto Answer Counter is to set after the ring count met the number you set then the auto answer will enable.

For security issue, You'd better to set the PIN Code. If you have set the PIN code, you will hear a tone to inform you input the PIN Code then you can dial out. Please notice that the PIN Code function couldn't function with Trunk Gateway function together.


Auto Answer
You could enable/disable the auto answer in this page.
Auto Answer: Off IP IN FXO IN Both Trunk Gateway
Auto Answer Counter: 3 (0~8)
PIN Code Enabled: Off On
PIN Code:
Submit Reset

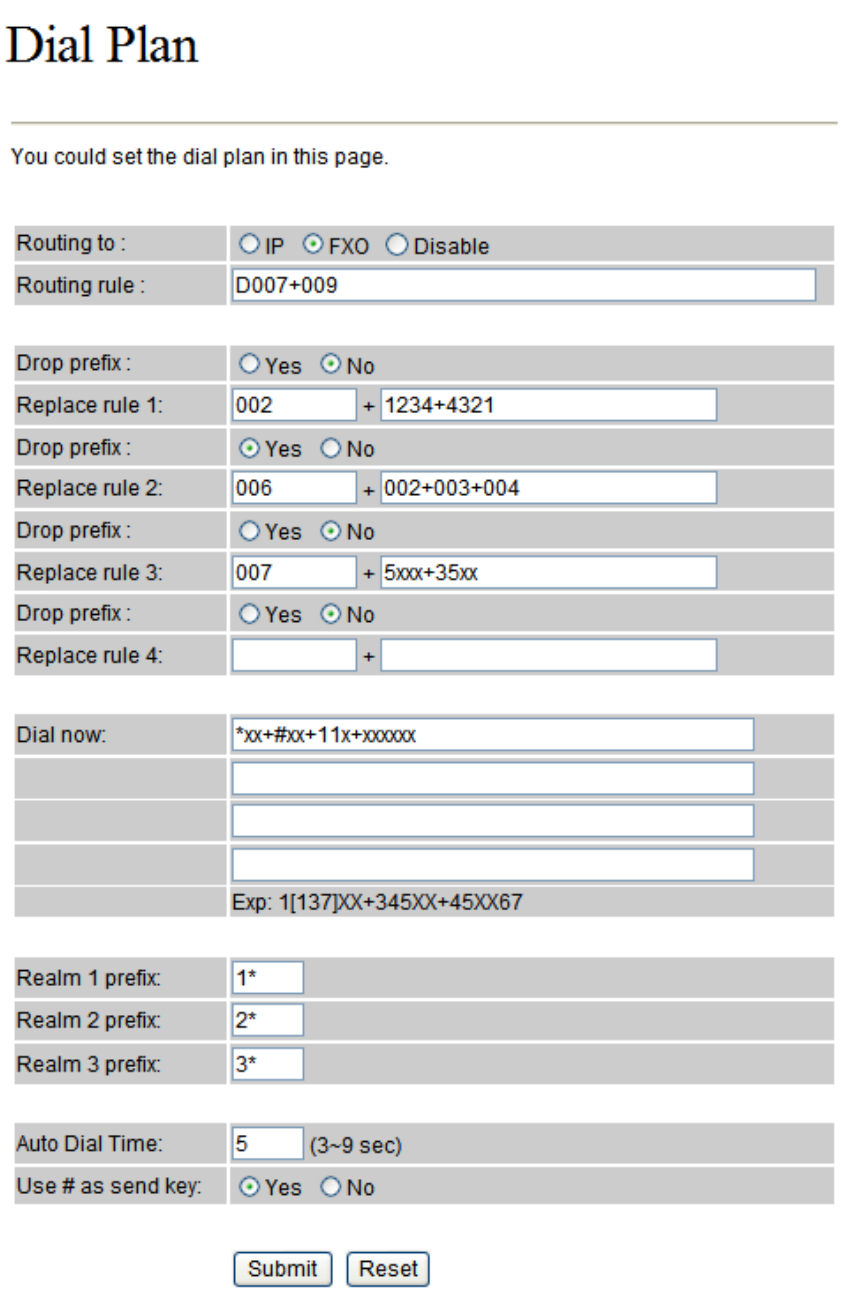
Caller ID settings

This page defines the device to show Caller ID in your PSTN Phone or IP Phone. There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.



Dial Plan Settings

This page defines the Dial Plan Setting function. This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.


Dial Plan
You could set the dial plan in this page.
Routing to : IP FXO Disable
Routing rule : D007+009
Drop prefix : Yes No
Replace rule 1: 002 + 1234+4321
Drop prefix : Yes No
Replace rule 2: 006 + 002+003+004
Drop prefix : Yes No
Replace rule 3: 007 + 5xxx+35xx
Drop prefix : Yes No
Replace rule 4: +
Dial now: *xx+#xx+11x+xxxxxx
Exp: 1[137]XX+345XX+45XX67
Realm 1 prefix: 1*
Realm 2 prefix: 2*
Realm 3 prefix: 3*
Auto Dial Time: 5 (3~9 sec)
Use # as send key: Yes No
Submit
Reset

Field	Description
Routing to	To select use IP or FXO types for auto routing function, and according to the Routing rule to handle the dialing numbers.
Routing rule	The rule can delete the prefix number.
Drop Prefix	The rule of add or replace code. If setup as No, it will add the prefix number prior to the identification number. If setup as Yes, it will replace the identification number.
Replace rule	The prefix number. It only accept the numeral and the max length is 8.
+	The identification number. It can accept the numeral or symbol and the max length is 40. - Symbol: It only accept the [+], [x] - +: It means as “or”. For example, [123+456+334+5xx] even if

	[123 or 456 or 334 or 5xx] - x: It is equal to 0~9. For example, [5xx] even if the number begin 5.
Dial Now	If the dialing number are match with this field, it will dial out and need not to press the "#" key to end the dialing. It accepts the numeral or symbol, and the max length are 124. ①**Note**: The starting number can't be the "0". For example, if the number is "0xxxx", because the starting number is "0", so that the system will ignore this dial plan.
Realm 1/2/3 prefix	These options can define the switching code for each Realm No.
Auto Dial Time	Stop dialing after seconds then send dial number out.
Use # as send key	If setup as Yes, the symtem sill stop to receive the dialing number when receive the [#] key. The system also will to determine the Auto Dial Time, it will carry out the calling if there isn't receive the digit after the Auto Dial Time. If setup as No, the system just according to the Auto Dial Time to determine the end time.

Descriptions of example:

Example_1: Routing to: **FXO**, Routing rule: **D007+009**

1. If the dialing number is "**00722199518**", it will delete 007 and send the following number for dialing out via FXO port. The real dialing number is [**22199518**].
2. If the dialing number is "**00922199518**", it won't delect any number. The real dialing number is [**00922199518**].

Example_2: Drop prefix: **No**, Replace rule 1: **002**, +: **1234+4321** (No limit the digit length)

3. If the dialing number is start as "**1234**", it will add the 002 at begin. The real dialing number is [**0021234…**].
4. If the dialing number is start as "**4321**", it will add the 002 at begin. The real dialing number is [**0024321…**].

Example_3: Drop prefix: **Yes**, Replace rule 2: **006**, +: **002+003+004** (No limit the digit length)

1. If the dialing number is start as "**002**", it will replace 002 by 006. The real dialing number is [**006…**].
2. If the dialing number is start as "**003**", it will replace 003 by 006. The real dialing number is [**003…**].

Example_4: Drop prefix: No, Replace rule 3: **007**, +: **5xxx+35xx** (Has limit the digit length)

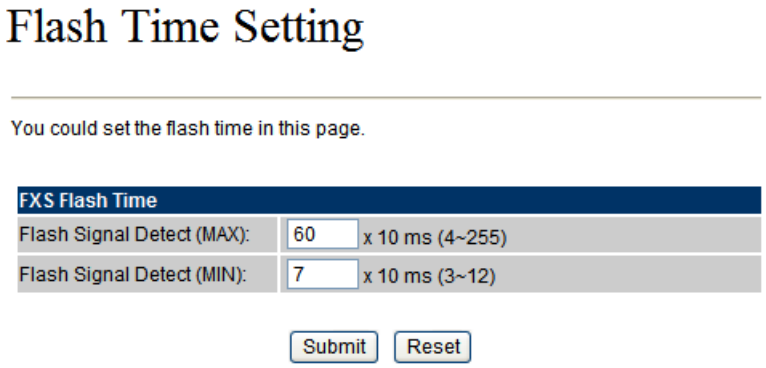
1. If the dialing number start as "**5**" and follow 3 digits, it will add the 007 at begin. The real dialing number is [**0075xxx**].
2. If the dialing number start as "**35**" and follow 2 digits, it will add the 007 at begin. The real dialing number is [**00735xx**].

Example_5: Dial Now: ***xx+#xx+11x+xxxxxx**

1. If the dialing number is match with the rule of "***xx**", it will send out the dialing number directly. For example, ***00/ *01/ *02…*99**.
2. If the dialing number is match with the rule of "**#xx**", it will send out the dialing number directly. For example, **#00/ #01/ #02…#99**.
3. If the dialing number is match with the rule of "**11x**", it will send out the dialing number directly. For example, **111/ 112/ 113…119**.
4. If the dialing number is match with the rule of 8 digits, it will send out the dialing number directly. For example, **12345678**.

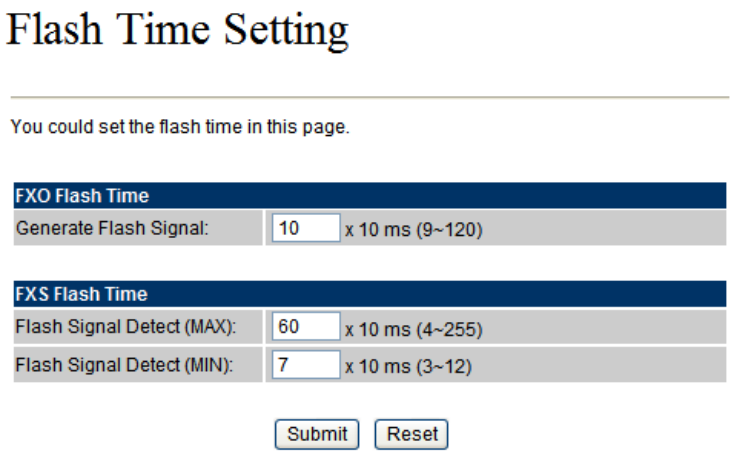
Flash Time Setting

When you use the PSTN Phone and you need to press the Hook to do the Flash (Switch to the other phone line or HOLD), this function is for you to set the time you press the Hook to represent the Flash function.


Flash Time Setting
You could set the flash time in this page.
FXS Flash Time
Flash Signal Detect (MAX): 60 x 10 ms (4~255)
Flash Signal Detect (MIN): 7 x 10 ms (3~12)
Submit
Reset

Flash Time Settings for VIP-156/VIP-156PT/VIP-157S

Beside the above settings, VIP-157 also can set the flash time of FXO port.


Flash Time Setting
You could set the flash time in this page.
FXO Flash Time
Generate Flash Signal: 10 x 10 ms (9~120)
FXS Flash Time
Flash Signal Detect (MAX): 60 x 10 ms (4~255)
Flash Signal Detect (MIN): 7 x 10 ms (3~12)
Submit
Reset

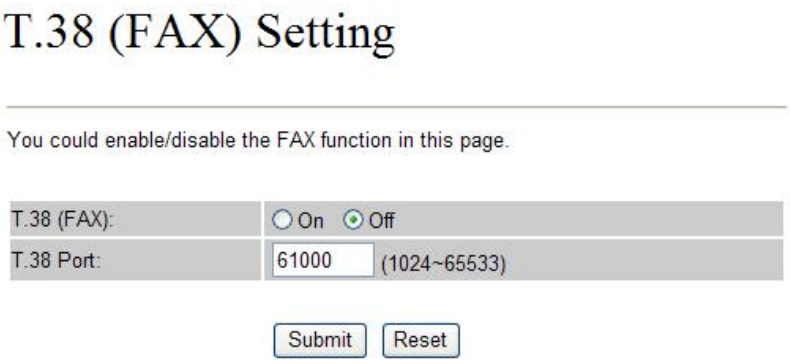
Flash Time Settings for VIP-157

Call waiting Settings

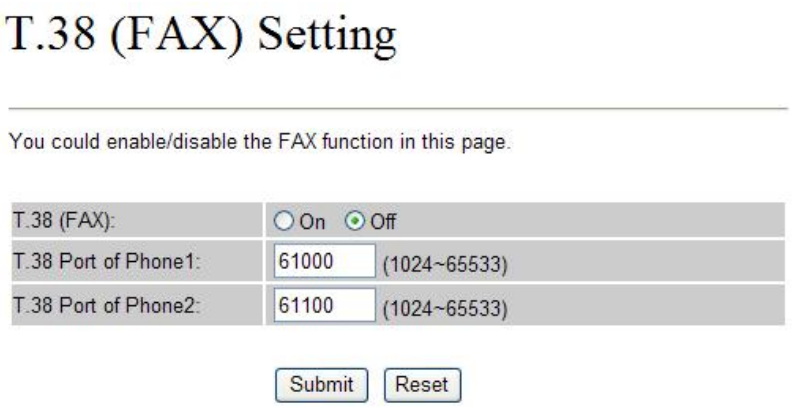
When you are talking with other people, You can choose If you want to hear the notice when there is a new coming call. If the call waiting function is On, if there is a new incomeing call, you will hear the call waiting notice in your current call. If you set the function to Off, then you will not hear any notice.

T.38 (FAX) Settings

This page defines the T.38 (FAX) Setting function. You can Enable/Disable the T.38 function, and can modify the T.38 transmission port of each FXS port.



T.38 (FAX) Settings for VIP-156/VIP-156PE/VIP-157



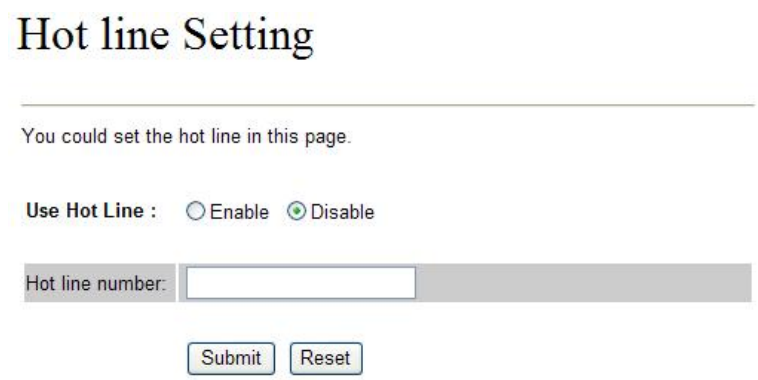
T.38 (FAX) Settings for VIP-157S

Hot line Settings

This page defines the Hot line setting in this page. When user pick up the handset, the device will call to the specific number automatically.

Use Hot Line: Click Enable to carry the Hot line function out.

Hot line number: The hot line number, it can input the IP address or registration number.



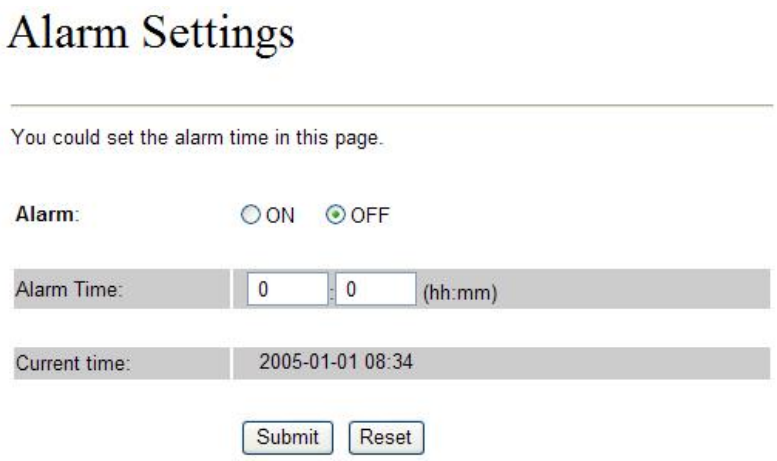
Alarm Settings

This page defines the Alarm setting in this page. It provides the alarm function, and it can set up the Alarm Time to get the telephone ringed up every day.

Alarm: The default is Off. If set up as On, the telephone will ringed up at the specific time.

Alarm Time: It can set up the system prompt time with 24 hours.

Current time: The next alarm time.



LAN Settings

This page defines the LAN setting in this page.

LAN Mode: The default is Bridge mode, and it also provides NAT mode.

- **Bridge:** When set as is mode, the LAN and PC ports are in the same network segment.
- **NAT:** The LAN and PC ports are in the different network segment, and PC port could enable the DHCP Server function to allot the IP address.

IP Type: The default is Fixed IP, and it also provides DHCP Client and PPPoE connection modes.

- **Fixed IP:** It could setup the IP address manual.
- **DHCP Client:** It will acquire the IP address automatically.

- **PPPoE:** It will use the PPPoE connection method.

IP: The IP address

Mask: The sub net address

Gateway: The default gateway address

DNS Server1: The default is 168.95.192.1, it could setup the first DNS server address.

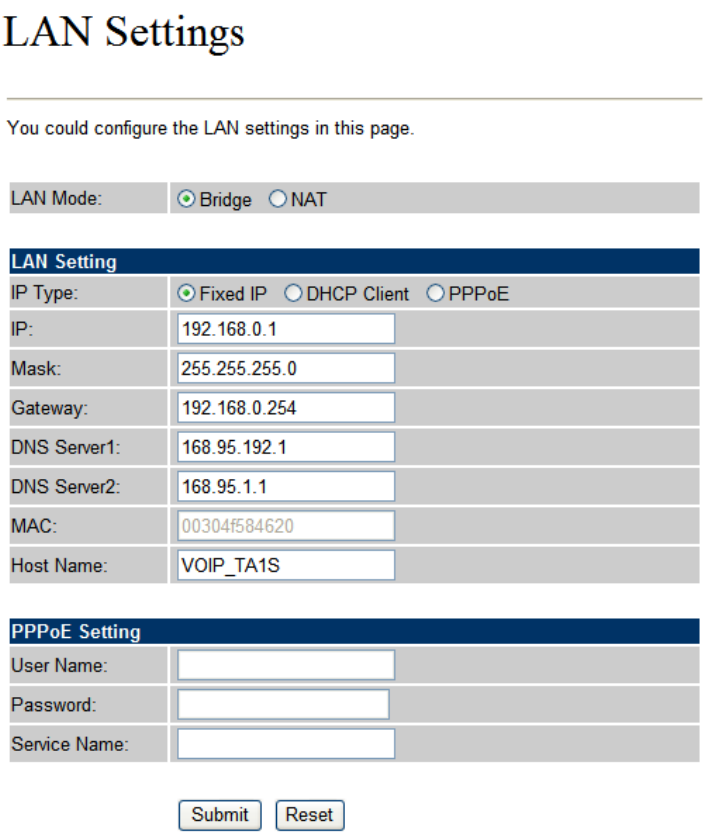
DNS Server2: The default is 168.95.1.1, it could setup the second DNS server address.

MAC: The MAC of LAN port

Host Name: The product model

User Name: The PPPoE connection account name. It could inpout numeral or character, the maximum date length are 63.

Password: The PPPoE connection account password. It could inpout numeral or character, the maximum date length are 63.



PC Settings

This page defines the PC setting in this page.

IP: The IP address of PC port. (**Default**: 192.168.0.1)

Mask: The sub net address. (**Default**: 255.255.255.0)

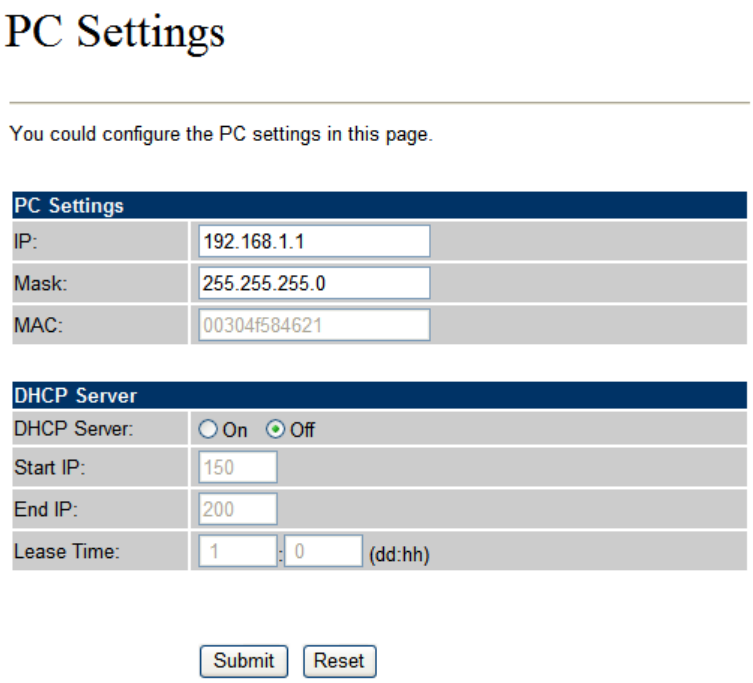
MAC: The MAC of PC port

DHCP Server: It will allot the IP address automatically when enabke this function.

Start IP: Start IP of lease table

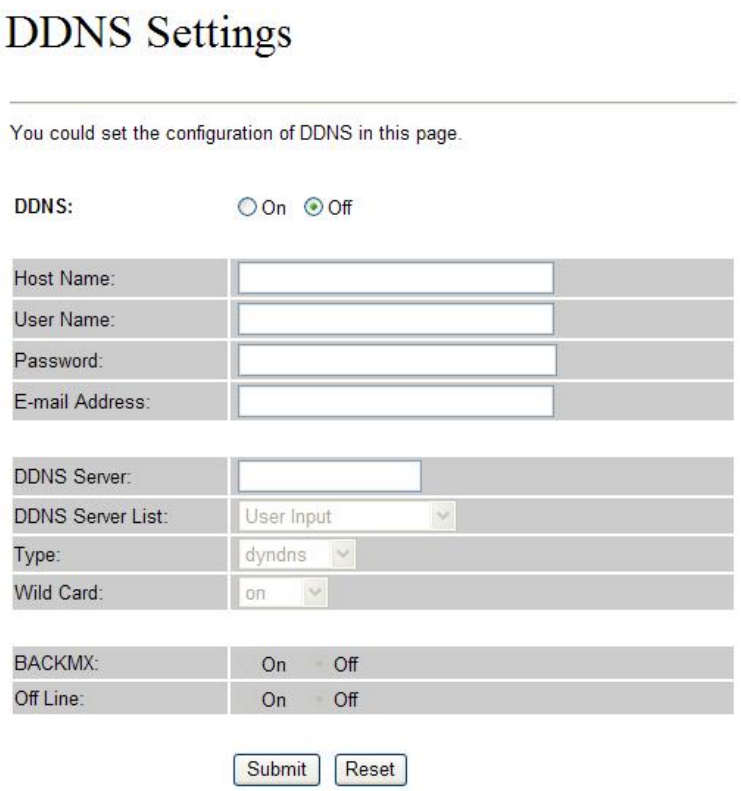
End IP: End IP of lease table. Network device connecting to the PC port can dynamic obtain the IP in the range between start IP and end IP

Lease Time: DHCP server lease time



DDNS Settings

This page defines the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.



VLAN Settings

This page defines the VLAN setting in this page. This function needs to co-operate with network devices which have VLAN function.

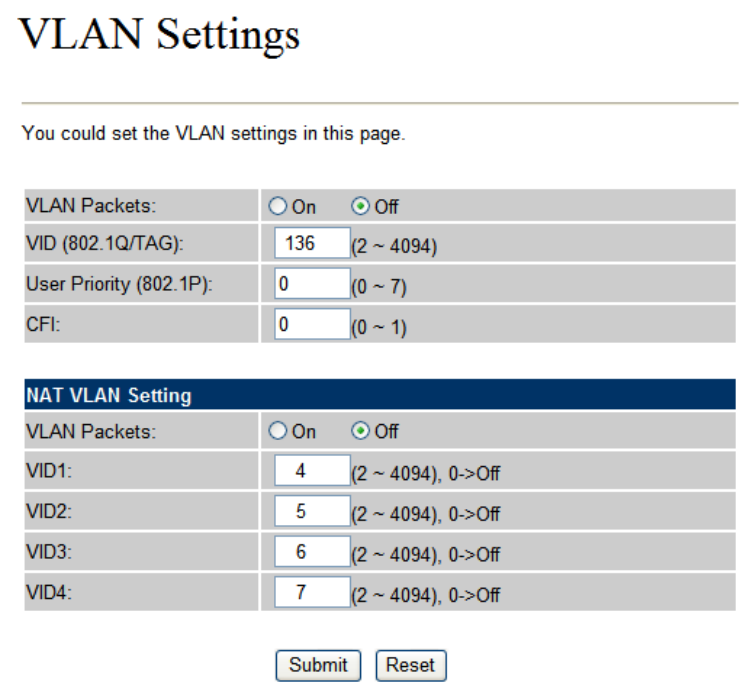
VLAN Packets: If setup as On, it could receive VLAN messages.

VID (802.1Q/TAG): Dispose VLAN ID is add a Tag header after realize enable the VLAN function. The realized voice packets transfer at the same VLAN. The prerequisite is it must the same as VLAN of upper switch. The value range are 2~4094.

User Priority (802.1P): To setup the user priority.

CFI: To indicate the Canonical Format.

- If CFI=1, it means the header label include RIF field, and the NCIF flag valus of RIF will to decide the MAC address is Canonical Format or Non-Canonical Format in frame information.
- If CFI=0, it means the header label does not include RIF field, and the MAC address is Canonical Format in frame information.

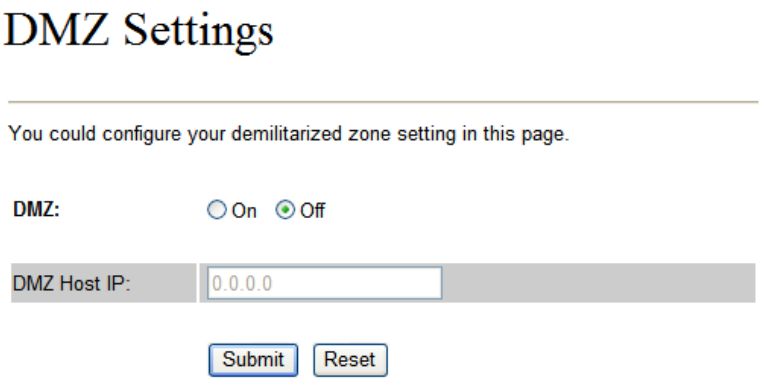

VLAN Settings
You could set the VLAN settings in this page.
VLAN Packets: On Off
VID (802.1Q/TAG): 136 (2 ~ 4094)
User Priority (802.1P): 0 (0 ~ 7)
CFI: 0 (0 ~ 1)
NAT VLAN Setting
VLAN Packets: On Off
VID1: 4 (2 ~ 4094), 0->Off
VID2: 5 (2 ~ 4094), 0->Off
VID3: 6 (2 ~ 4094), 0->Off
VID4: 7 (2 ~ 4094), 0->Off
Submit
Reset

DMZ Settings

This page defines the DMZ setting in this page.

DMZ: If setup as On, all of packets (expect SIP packets) will send to the specific IP address.

DMZ Host IP: The DMZ host IP address.



Virtual Server

This page defines the Virtual Server setting in this page. You could define 24 virtual service information in this page. When you finished the setting, please click the Submit button.

Virtual Server Page: There are total page1 to page 3. It could choose the page which want to go over.

Num: The serial number. There are total 24 records from Num 0 to 23.

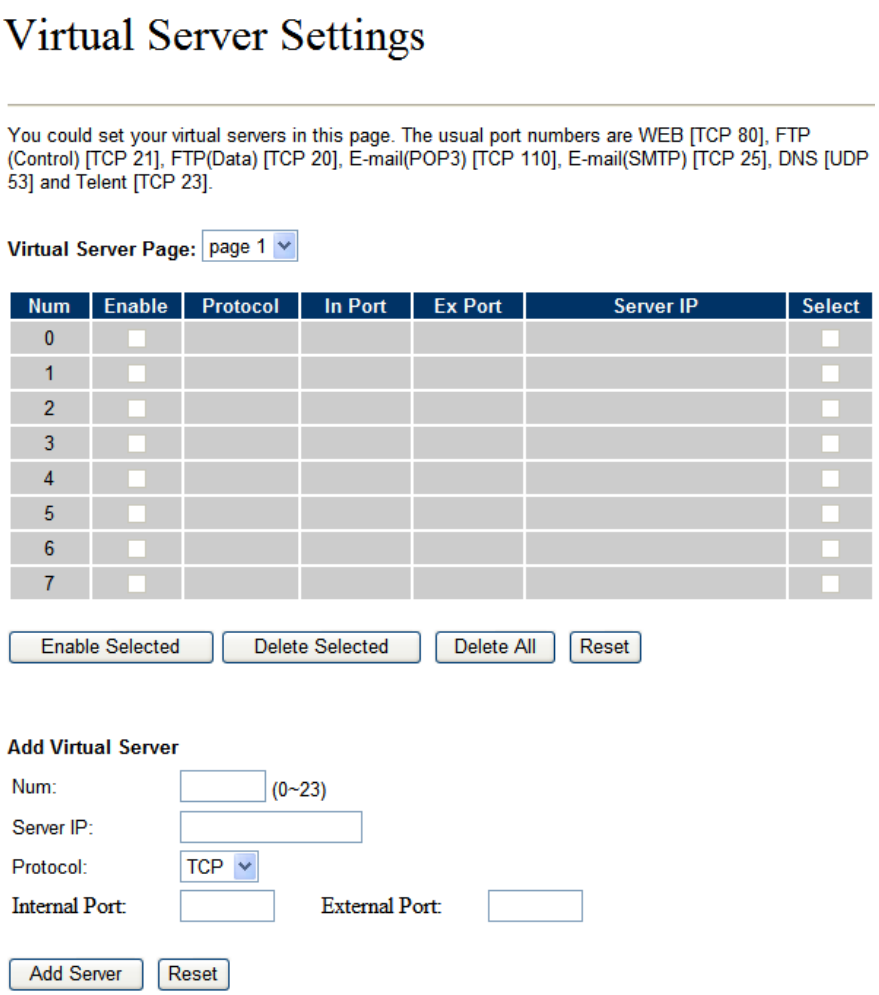
Enable: The activate status. The default is Disable, this record will been activate if enable.

Protocol: The TCP or UDP communication protocol.

Internal Port: For corresponding the internal port.

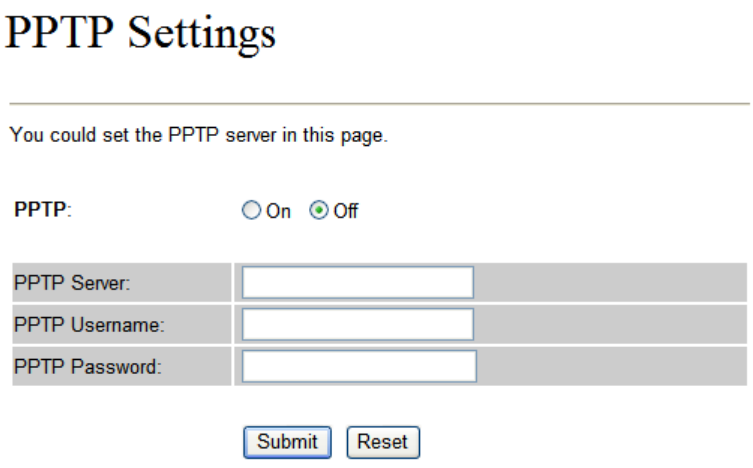
External Port: For corresponding the external port.

Server IP: To input the Server IP address.


Virtual Server Settings
You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].
Virtual Server Page: page 1
Num Enable Protocol In Port Ex Port Server IP Select
0 1 2 3 4 5 6 7
Enable Selected
Delete Selected
Delete All
Reset
Add Virtual Server
Num: (0~23)
Server IP:
Protocol: TCP
Internal Port:
External Port:
Add Server
Reset

PPTP Settings

This page defines the PPTP setting in this page. You could setup the PPTP Server connection information. When you finished the setting, please click the Submit button.



Service Domain Settings

In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the ATA. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

Display Name: you can input the name you want to display.

Line number: you need to input the User Name get from your ISP.

Register Name: you need to input the Register Name get from your ISP.

Register Password: you need to input the Register Password get from your ISP.

Domain Server: you need to input the Domain Server get from your ISP.

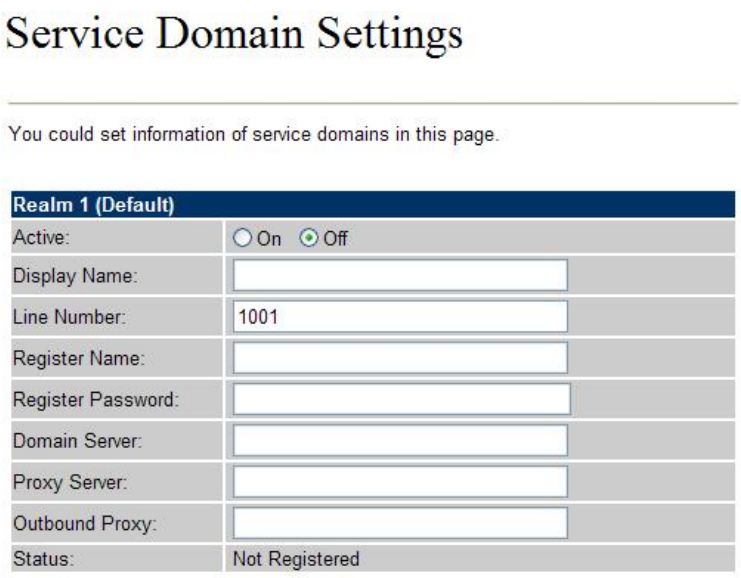
Proxy Server: you need to input the Proxy Server get from your ISP.

Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

You can see the Register Status in the Status item. If the item shows “Registered”, then your Phone Adapter is registered to the ISP, you can make a phone call direcly.

If you have more than one SIP account, you can following the steps to register to the other ISP.

When you finished the setting, please click the Submit button.



ⓘNote:

ATA can register to three different SIP Proxies at the same time. It can receive any one of different SIP accounts incoming call, and it can switch to any one SIP accounts for making calls through input the switch code.

Realm switch code:

1*: Realm 1

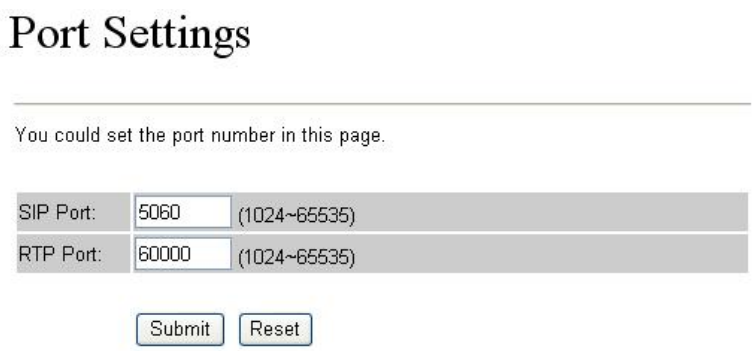
2*: Realm 2

3*: Realm 3

For example: The default is realm 1, input the **2*** (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

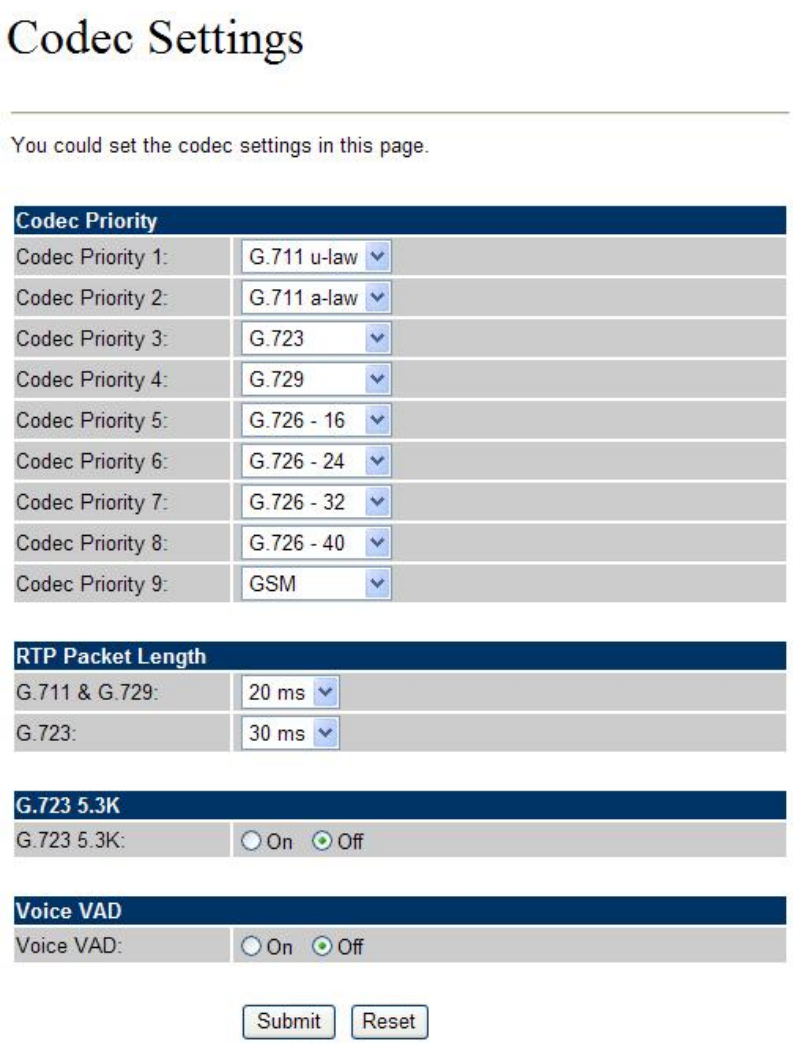
Port Settings

This page defines the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.



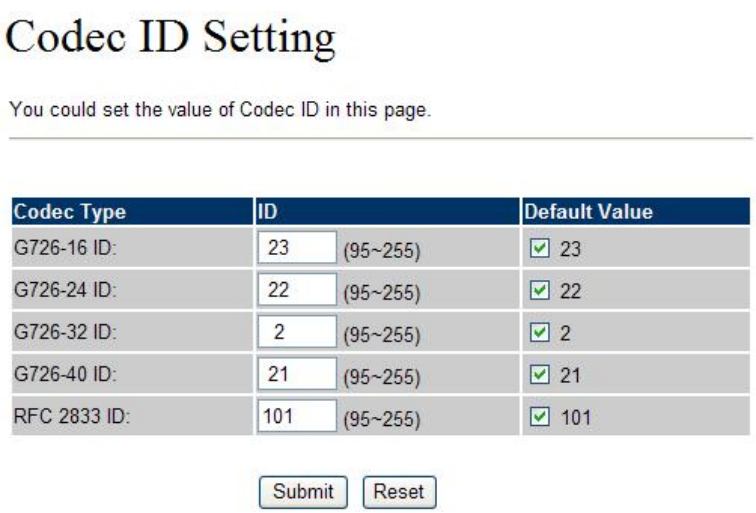
Codec Settings

This page defines the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.


Codec Settings
You could set the codec settings in this page.
Codec Priority
Codec Priority 1: G.711 u-law
Codec Priority 2: G.711 a-law
Codec Priority 3: G.723
Codec Priority 4: G.729
Codec Priority 5: G.726 - 16
Codec Priority 6: G.726 - 24
Codec Priority 7: G.726 - 32
Codec Priority 8: G.726 - 40
Codec Priority 9: GSM
RTP Packet Length
G.711 & G.729: 20 ms
G.723: 30 ms
G.723 5.3K
G.723 5.3K: On Off
Voice VAD
Voice VAD: On Off
Submit
Reset

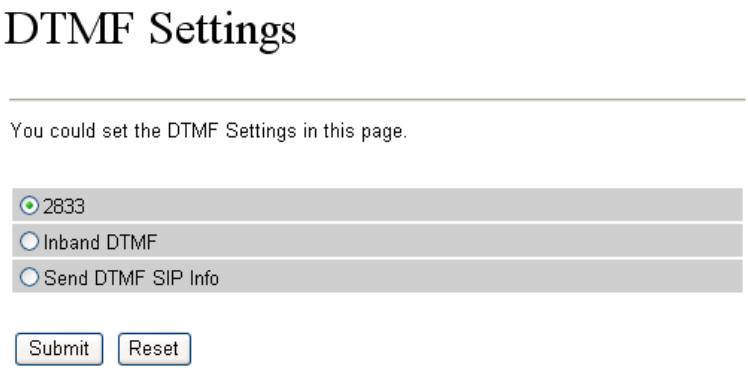
Codec ID Setting

This page defines the Codec ID. Sometimes 2 VoIP device with different Codec ID will cause the interopability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, then you can change your Codec ID. When you finished the setting, please click the Submit button.



DTMF Setting

This page defines the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.



RPort Settings

This page defines the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

Other Settings

This page defines the Hold by RFC, Voice/SIP QoS and other settings in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

Hold by RFC: The default is disable, and to start up communication hold back function (RFC definition). Set enable to start up the Hold by RFC function.

Voice QoS (Diff-Serv): The Voice QoS feature.

SIP QoS (Diff-Serv): The SIP QoS feature.

The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

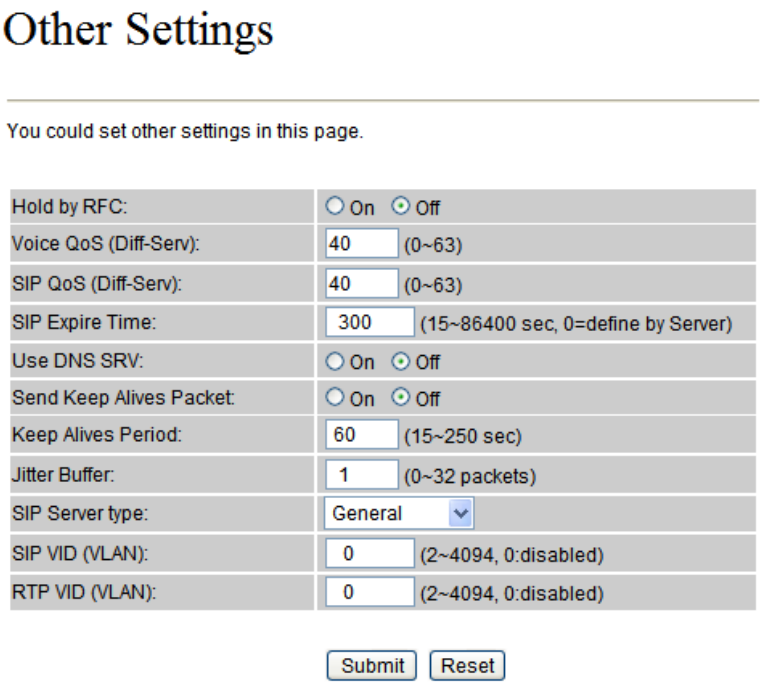
SIP Expire Time: To setup the registration interval time.

Use DNS SRV: The default is disable, and use DNS SRV mode. Set enable to use DNS to SRV mode to search the host information.

Send Keep Alive Packet: Always to transport the network packets to keep the NAT port could be opened continuous.

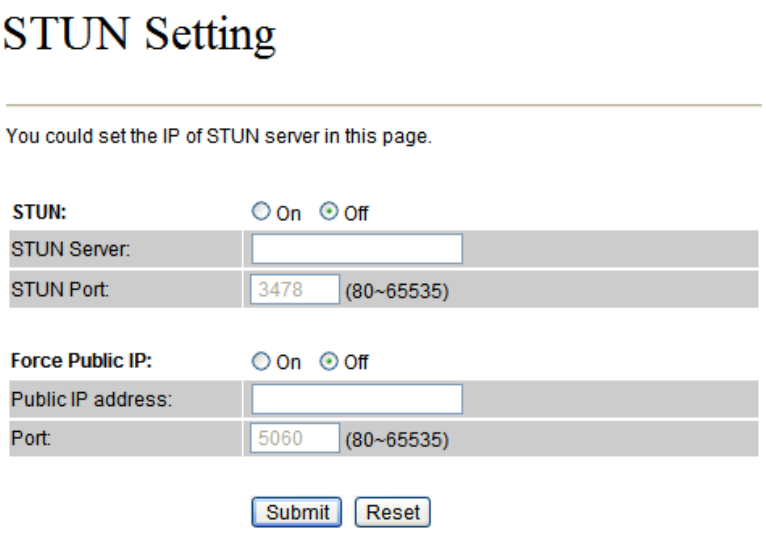
Keep Alive Period: To setup the interval time for transporting packets.

Jitter Buffer: To setup the size for jitter buffer packets.


Other Settings
You could set other settings in this page.
Hold by RFC: On Off
Voice QoS (Diff-Serv): 40 (0~63)
SIP QoS (Diff-Serv): 40 (0~63)
SIP Expire Time: 300 (15~86400 sec, 0=define by Server)
Use DNS SRV: On Off
Send Keep Alives Packet: On Off
Keep Alives Period: 60 (15~250 sec)
Jitter Buffer: 1 (0~32 packets)
SIP Server type: General
SIP VID (VLAN): 0 (2~4094, 0:disabled)
RTP VID (VLAN): 0 (2~4094, 0:disabled)
Submit
Reset

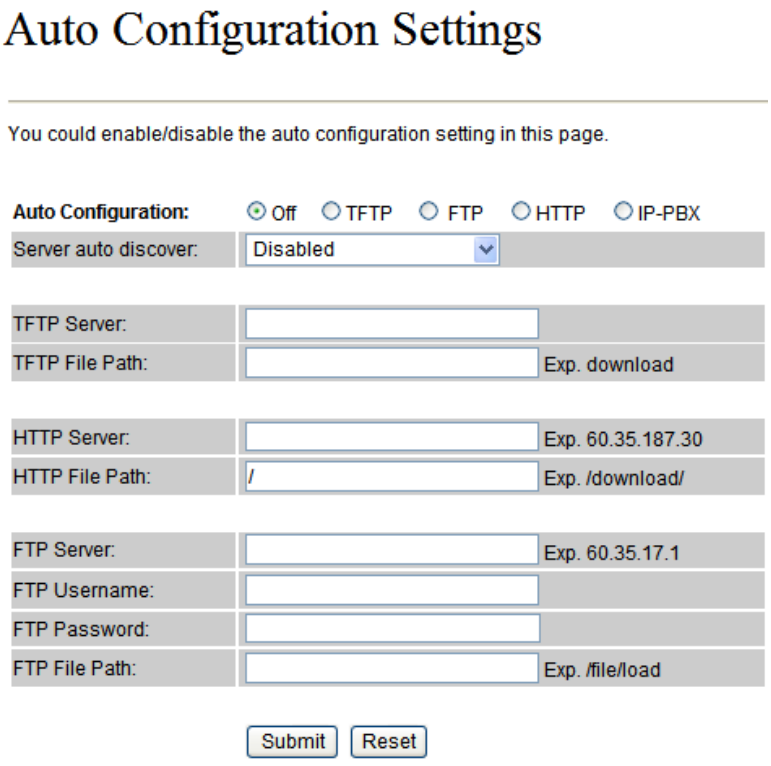
STUN settings

This page defines the STUN Enable/Disable and STUN Server IP address in this page. This function can help your Phone Adapter working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.



Auto Configuration

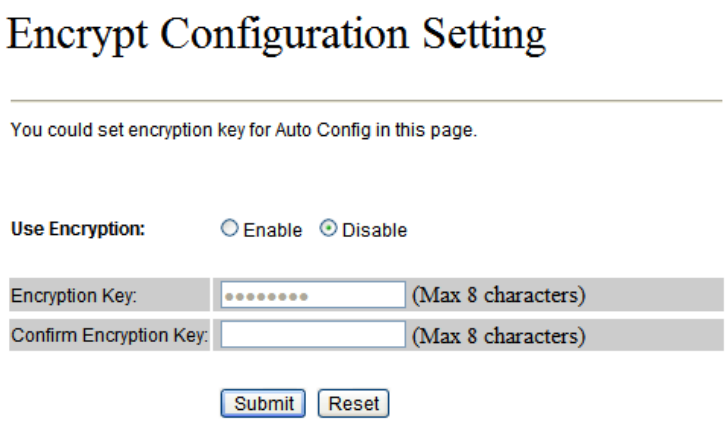
This page defines the Auto Configuration (Auto Provision) setting. ATA supports TFTP, FTP, HTTP and IP PBX auto configuration function in total. In IP PBX Auto Configuration Setting you need to check with your service provider if they have provided this function. Usually this function will be boundle with an IP PBX to use in the office.



Encrypt Config

This page can decide if use encryption key for Auto Configuration (Auto Provision) funciton. The encryption key needs be the same as config file for Auto Configuration. If the encryption key is wrong , ATA won't to write in the settings when got the config file.

If ATA cancel encryption key, it won't need to input encryption key when generating the config file.

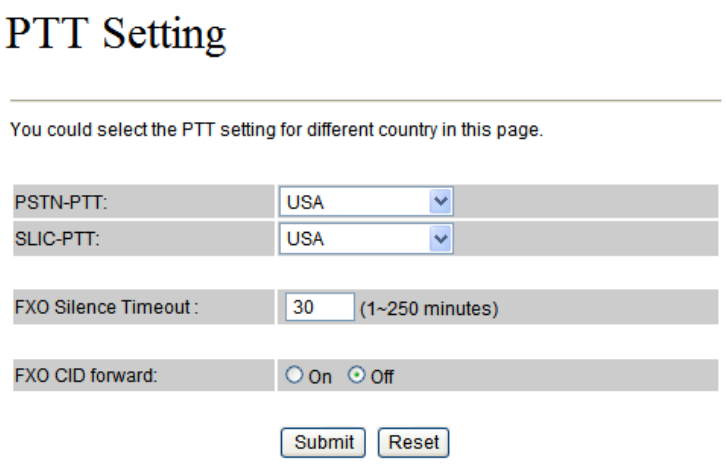


PTT Settings

In PTT Settings is for you to set the Country, different country will have different settings in FXS inter face.

PTT Settings for VIP-156/VIP-157S

Beside the above settings, VIP-157 also can set country of FXO port.



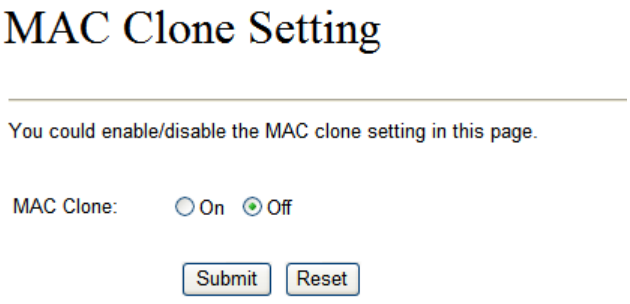
PTT Settings for VIP-157

Cancel without to tag

This function can decide the device if send the cancel tag to Proxy Server. If there has the similar symptom that the caller cancel the call before the called answer the call, the called still continue to ring up even the caller has cancel this call. It could try select this function to **Yes** to avoid the above symptom.

MAC Clone Setting

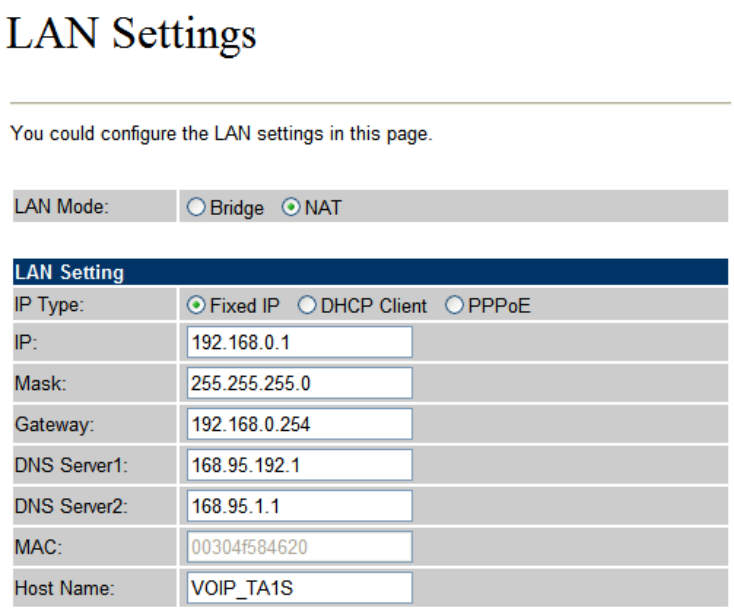
This page defines the MAC Clone Enable/Disable. This function will copy the MAC address from NIC (Network Interface Card) which placed in PC to LAN port of ATA. That because some ISP will limit the MAC address for PPPoE dial-up connection.



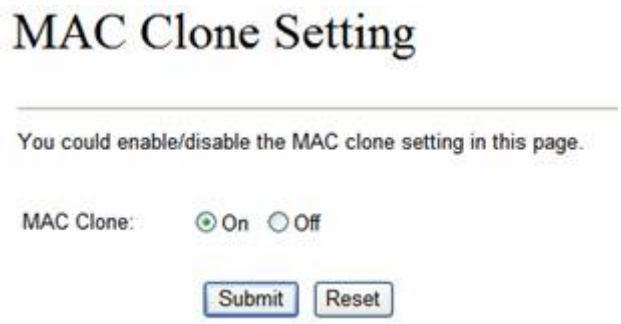
Please refer to the following operate procedures for more understandings to carry out the MAC Clone

function.

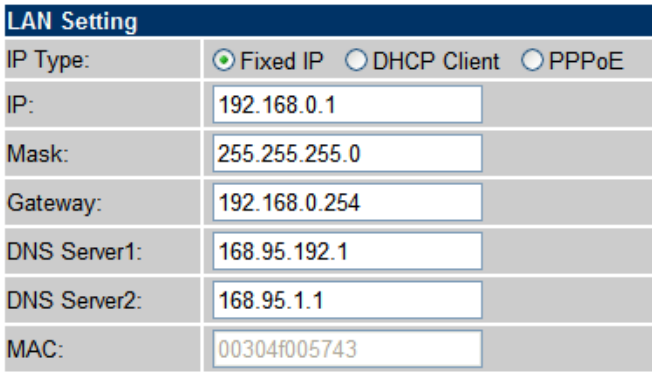
1. Please login ATA and browse to “**Network -> LAN Settings**” page. To switch the LAN mode to **NAT** mode then press **Save&Reboot** button to save the settings and reboot machine.



2. Please make sure the network cable of your PC directly connect with PC port of ATA, then re-login ATA. (The default IP address of PC port is http://192.168.123.1)
3. Please browse to “**Advanced Settings -> MAC Clone Setting**” page and enable the MAC Clone function.

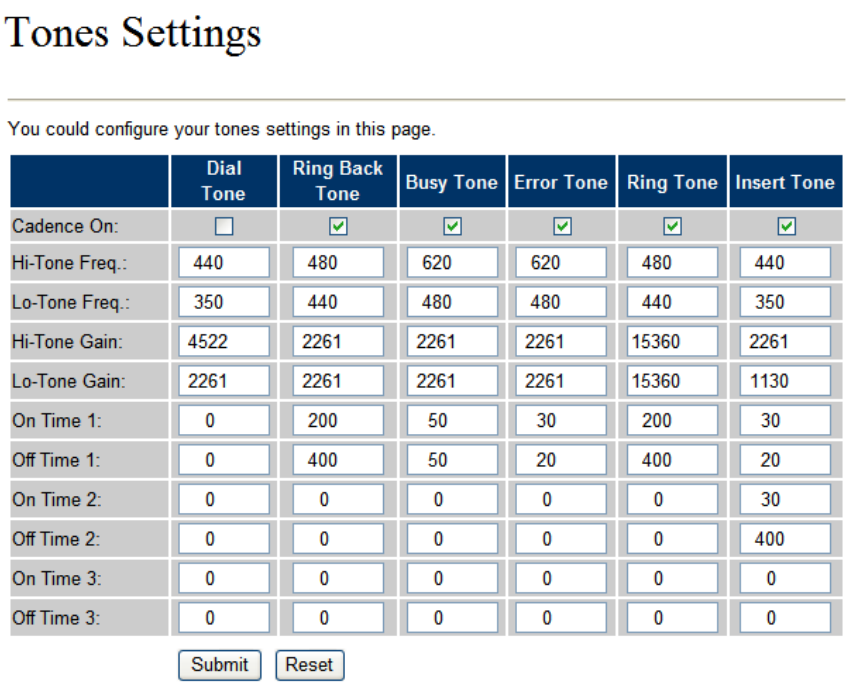


4. ATA will prompt if sure want to clone the MAC of your PC to the LAN port of ATA.
5. After **Save&Reboot**, the MAC of LAN port will become to PC’s original MAC address.



Tone Settings

This page defines the Tone settings. This function can setup the related parameters of Dial Tone, Ring Back Tone, Busy Tone, Error Tone and Inser Tone. When you finished the setting, please click the Submit button.



Tones Settings

You could configure your tones settings in this page.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	☐	☑	☑	☑	☑	☑
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

Submit Reset

Advanced Settings

This page defines the advanced functions. When you finished the setting, please click the Submit button.

ICMP Not Echo: This function can disable echo when someone ping this device, it can avoid haker try to attack the device.

Send Anonymous CID: If enable this function, machine will to start the calling hidden function, and it will not send the related Caller information. (The Registration Server also need support this function)

Billing Signal: There are provide three type billing types: Polarity Reversal, Tone_12K and Tone_16K. (The Registration Server also need support this function)

CPC Delay: When receive the disconnect signal, machine will to cut the voltage down to 0V after this time.

CPC Duration: When starting to cut the voltage down to 0V, machine will to continue this state by this time.

Send Flash event: There are provide two flash formats: DTMF Event and SIP Info.

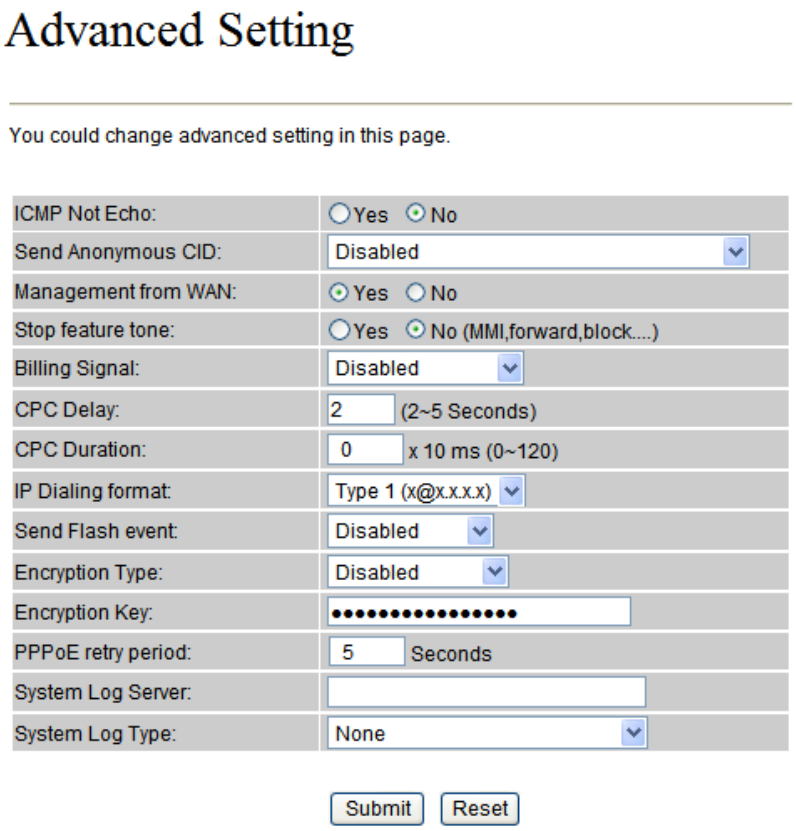
SIP Encrypt: There are provide seven encrypt formats: INFINET, AVS, WALKERSUN1, WALKERSUN2, CSF1, CSF2 and GX. (The Registration Server also need support this function)

PPPoE retry period: If PPPoE dial-up connection fail, machine will retry the dial-up motion after this

time.

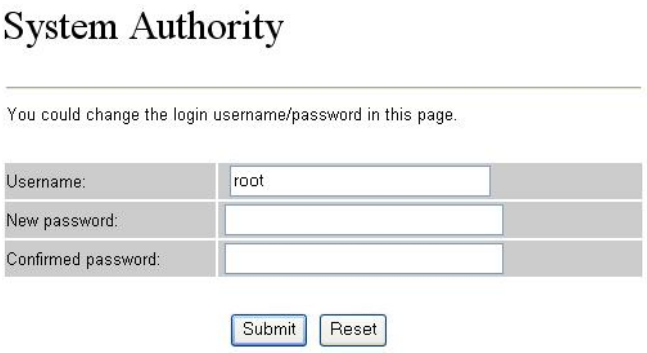
System Log Server: Machine could send the system logs to the specific Syslog Server. It can input the IP or Domain address.

System Log Type: There are seven Syslog types: Call Statistics, General Debug, Call Statistics + General Debug, SIP Debug, Call Statistics + SIP Debug, General Debug + SIP Debug and All.


Advanced Setting
You could change advanced setting in this page.
ICMP Not Echo: Yes No
Send Anonymous CID: Disabled
Management from WAN: Yes No
Stop feature tone: Yes No (MMI,forward,block....)
Billing Signal: Disabled
CPC Delay: 2 (2~5 Seconds)
CPC Duration: 0 x 10 ms (0~120)
IP Dialing format: Type 1 (x@x.x.x.x)
Send Flash event: Disabled
Encryption Type: Disabled
Encryption Key: ••••••••••••••••
PPPoE retry period: 5 Seconds
System Log Server:
System Log Type: None
Submit Reset

System Authority

In System Authority you can change your login password.



Save & Reboot

In Save & Reboot you can save the changes you have done. If you want to use new setting in the Phone Adapter, You have to click the Save button. After you click the Save button, the Phone Adapter

will automatically restart and the new setting will effect.

Firmware Upgrade

In Firmware Upgrade function you can update new firmware via HTTP method in this page. You can ugrade the firmware by the following steps:

Select the upgrade method and the firmware code type, CPU (AP) or DSP code.

Click the “Browse” button in the right side of the File Location or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the device then click the Update button.

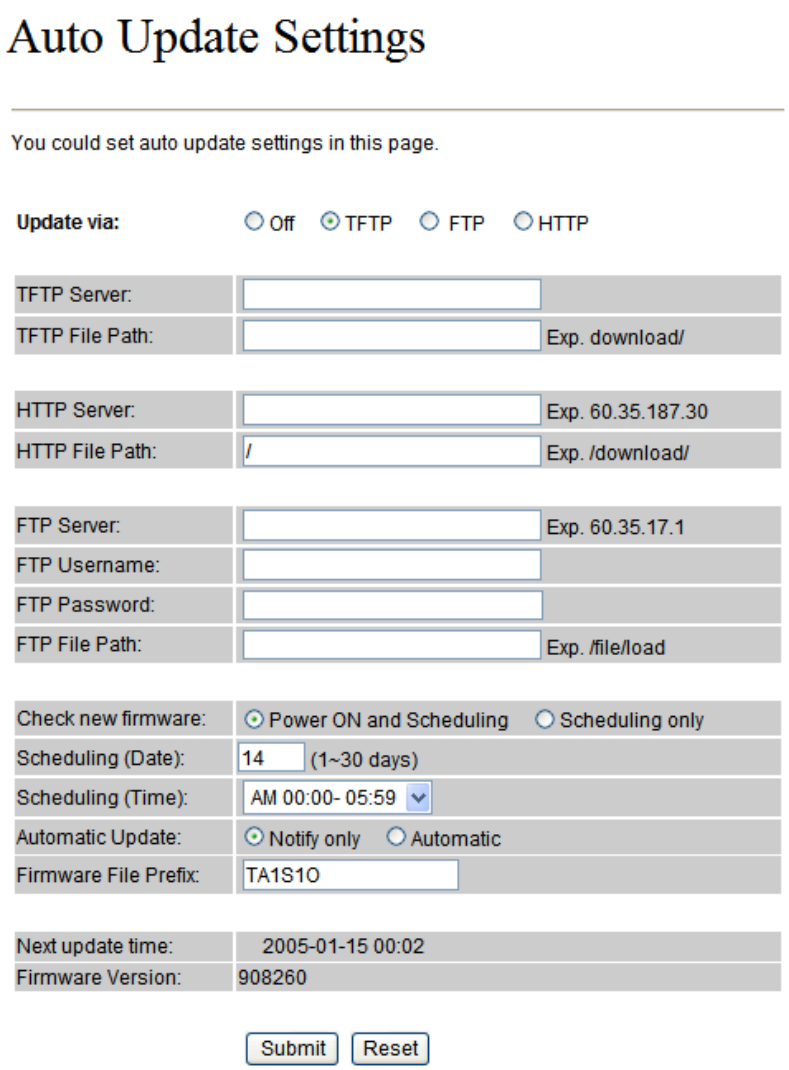
ⓘNote:

For technological consideration, we’ve strongly suggested refering to the following upgrade methods for update your device.

After firmware loaded, the unit will be reboot, and Default IP address of the customized firmware: http://192.168.0.1; login name/password: **root/null (no password)**

Auto Upgrade

The device can update new firmware with the **gz** or **ds** file format automatically by the Auto Upgrade function.



Field	Descriptions
Update via	There are TFTP/ FTP and HTTP three ways to provide the auto upgrade function.
TFTP Server	Input the TFTP Server address, and it could input the IP or Domain Name form.
TFTP File Path	Set up the file path.
HTTP Server	Input the HTTP Server address, and it could input the IP or Domain Name form.
HTTP File Path	Set up the file path.
FTP Server	Input the FTP Server address, and it could input the IP or Domain Name form.
FTP Username	The login username.
FTP Password	The login password
FTP File Path	Set up the file path.
Check new firmware	The device will according to the below ways to check the new

	firmware. **- Power On (+ Scheduling)**: The machine will check the new firmware when power on and following the scheduling date and time. **- Scheduling**: The machine will follow the scheduling date and time to check the new firmware.
Scheduling (Date)	The machine will check the new firmware between the time range by random.
Automatic Update	There are Notify only and Automatic ways to update. **- Notify only**: If there are new firmware, the ATA will send the "Be Be Be" sounds when pick up the handset to prompt there are new firmware. **- Automatic**: The device will carry firmware update out automatically.
Firmware File Prefix	It will check the information of model name.
Next update time	It will show the next check date and time.

ⓘNote:

If the **Check new firmware** field selected to Power On, the machine will chck the new firmware accoeding the scheduling time/date and power on. If there are new firmware can be upgraded, the machine won't carry firmware update out automatic. The machine will send the prompt sounds when pick up the handset, and it needs to update firmware by manual.

Reset to Default

In Default Setting you can restore the Phone Adapter to factory default in this page. You can just click the Restore button, then the Phone Adapter will restore to default and automatically restart again.

Reboot without saving

Reboot function you can restart the Phone Adapter. If you want to restart the Phone Adapter, you can just click the Reboot button, then the Phone Adapter will automatically.

Reboot without Saving

You could press the reboot button to restart the system.

Reboot without Saving: Reboot

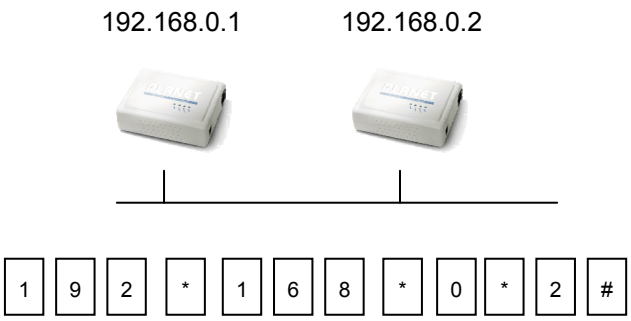
Appendix A Voice Communication Samples

There are several ways to make calls to desired destination in ATA. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

Case 1: ATA to ATA connection via IP address

Assume there are two ATAs in the network the IP address are 192.168.0.1, 192.168.0.2

Analog telephone sets are connected to the **phone** (RJ-11) port of ATAs respectively



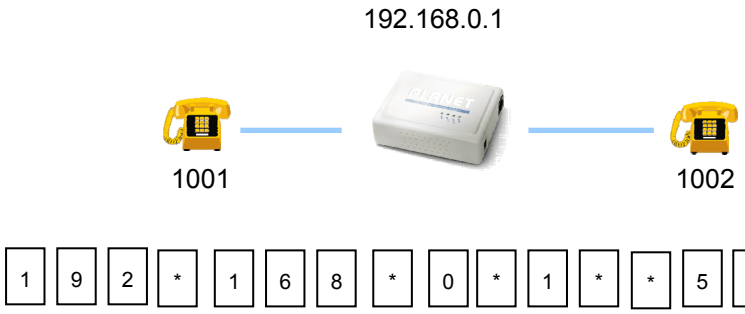
Test the scenario:

1. Pick up the telephone set on ATA A.
2. Press the keypad: **192*168*0*2#** shall be able to connect to the ATA B.
3. Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

Case 2: (Peer-to-Peer mode) VIP-157S Port 1 to Port 2 communications

Supposing one VIP-157S connects to two telephones, just pick up phone 1 and dial '**192*168*0*1**5062**', phone 2 will ring.

Analog telephone sets are connected to the phone (RJ-11) ports of VIP-157S respectively



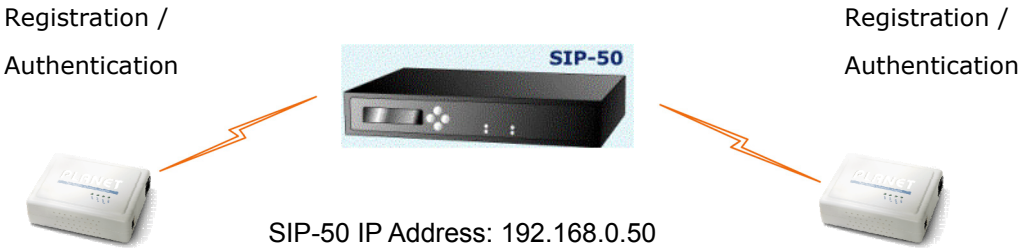
Test the scenario:

1. Pick up the telephone set on VIP-157S port 1, and you should be able to hear the dial-tone
2. Press the keypad: **192*168*0*1**5062#** shall be able to connect to the VIP-157S port 2
3. Then the telephone set in VIP-157S port 2 should ring. Please repeat the same dialing steps on port 2 to establish the first voice communication from VIP-157S

ⓘ Hint

- If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with a "**#**".
- If the ATAs are installed behind a NAT/firewall/IP sharing device for Peer-to-Peer VoIP application, please make sure the NAT device support SIP applications, and suitable settings should be applied to the NAT device to enable the SIP communications before making calls
- [**VIP-157S**] in PLANET ATA series products, to connect to remote ATA, press the keypad in the following sequence to connect to the remote VIP-157S port 2: **[remote ATA IP address]**5062**, for example: **192*168*0*2**5062**

Case 3: Voice communication via SIP proxy server – SIP-50



- ATA A IP Address: 192.168.0.1
  Line Number: 1001
- ATA B IP Address: 192.168.0.2
  Line Number: 2002

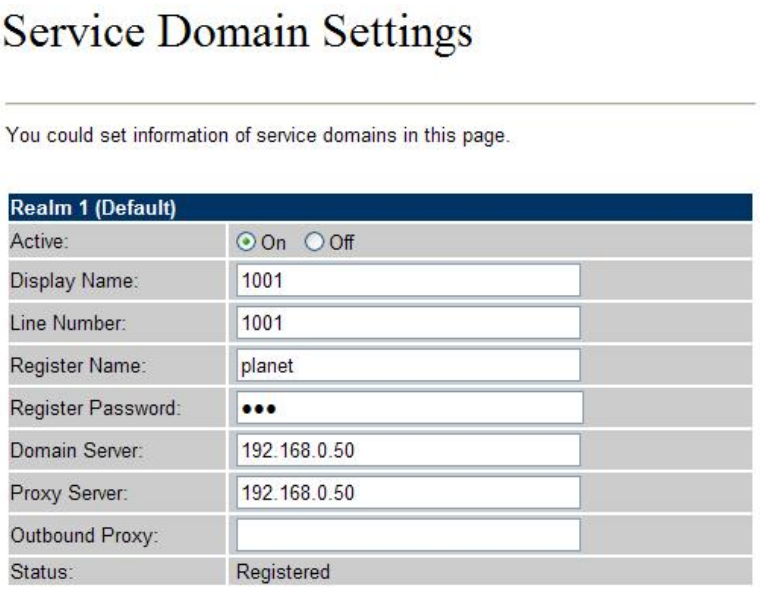
Device configurations on the ATA:

STEP 1:

Log in SIP-50 and create two testing accounts/password: **1001** / **123** (for ATA A), and **1002** / **123** (for ATA B) for the voice calls.

STEP 2:

Please log in ATA via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:



STEP 3:

Repeat the same configuration steps on ATA B, and check the machine registration status, make sure the registrations are completed.

Test the scenario:

1. Pick up the telephone on ATA A
2. Press the keypad: **2002** shall be able to connect to the ATA B
3. Then the telephone set in ATA B should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

Case 4: Voice communication via IP PBX system – IPX-2000 (Auto-config)

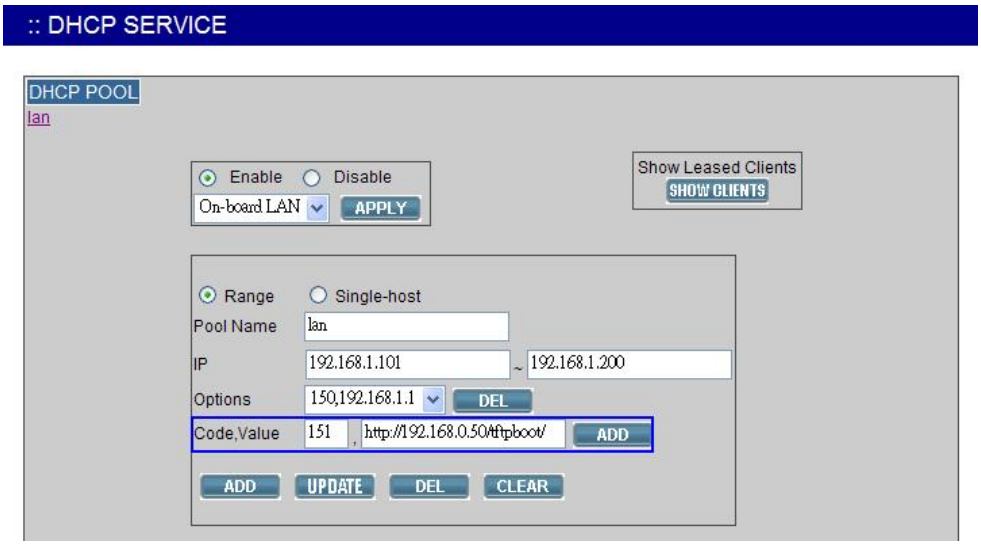
In the following sample, we'll introduce how to integrate the ATA with our IP PBX system IPX-2000 via Auto-config feature.

- ATA A IP Address: 192.168.0.1
  Line Number: 1001
- ATA B IP Address: 192.168.0.2
  Line Number: 2002

Device configurations on the IPX-2000:

STEP 1:

Log in IPX-2000 and browse to the **Srevice → DHCP Service** menu and create new options list for the auto configuration.



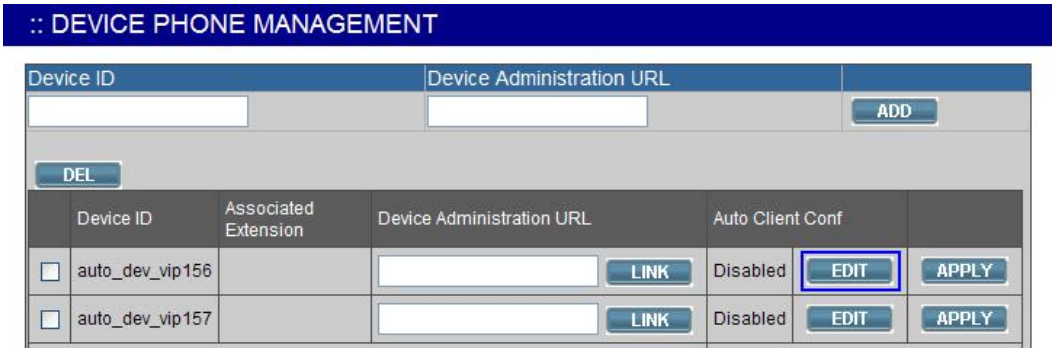
Code: please insert **151** as the DHCP server option.

Value: http://LAN IP for IPX-2000/tftpboot/

An example option 151 would be option=**151** value= **http://192.168.0.50/tftpboot/**

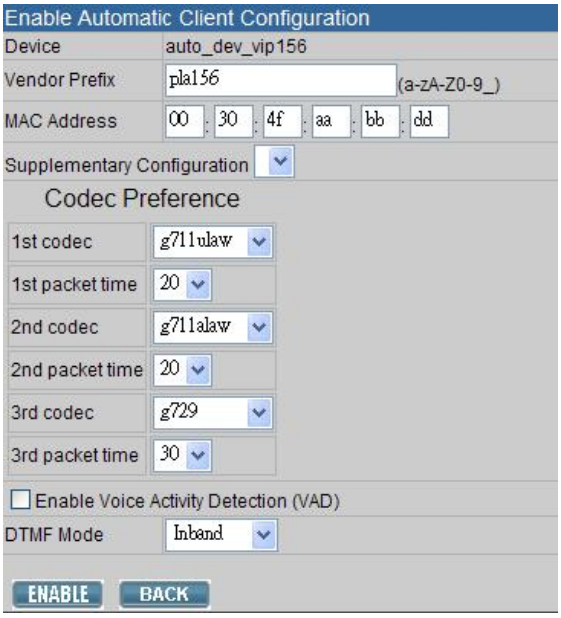
STEP 2:

Please browse to the **Device → IP Phone** menu and create new device. And press the **EDIT** button for set up the Auto Config configuration.



STEP 3:

Please fill out the Vendor Prefix code and MAC Address of ATA.



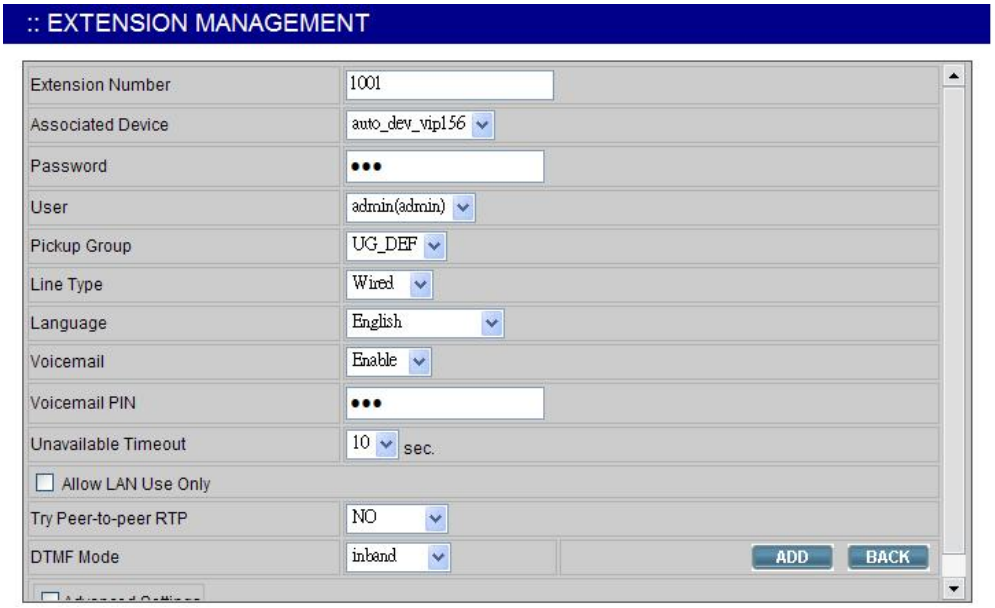
ⓘNote:

The following are the Vendor Prefix of devices:

1. VIP-156: **pla156**
2. VIP-157/VIP-157S: **pla157**

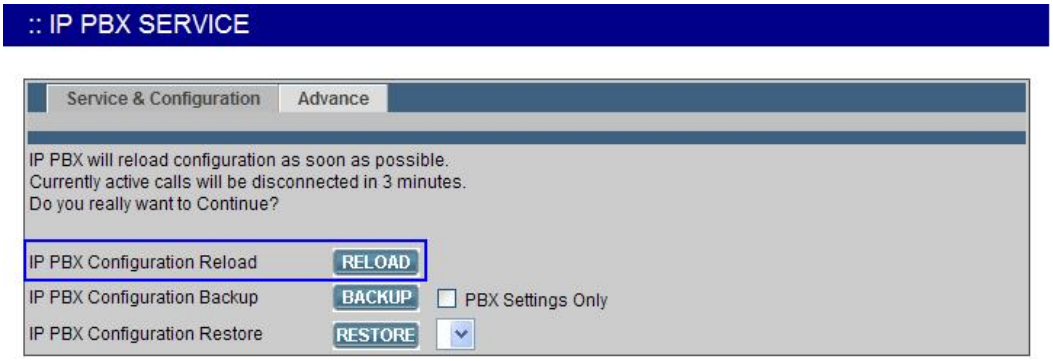
STEP 4:

Please browse to the **Device → Extension of IP Phone** menu to create the two extension accounts/password: **1001/123** (for ATA A), and **1002/123**(for ATA B) for the voice calls.



STEP 5:

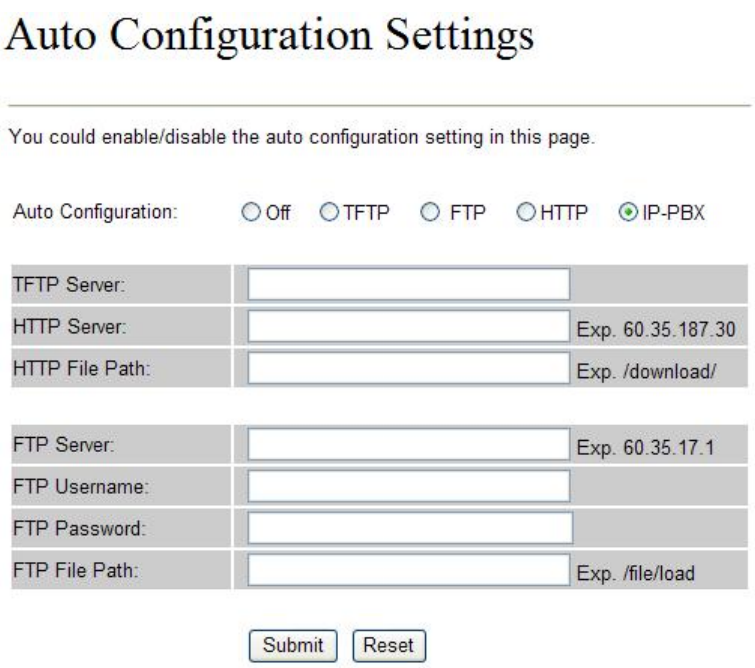
After setting up the parameters, please browse to the **Service → IP PBX service** menu, and press **RELOAD** of IP PBX configuration reload selection for activating the settings.



Device configurations on the ATA:

STEP 6:

Please log in ATA via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please browse to the Auto-config page, and enable the Auto Configuration features for IP PBX system.



STEP 7:

After enabling the auto-config feature, the ATA shall be able to obtain IP address and SIP extension information from IP PBX system IPX-2000 information. To verify the auto-config results, you may connect telephone set to ATA; press **#120#** to check if the IP address is obtained from IPX-2000. And **#122#** can be used to verify the extension number assigned by IPX-2000.

STEP 8:

Repeat the same configuration steps on ATA B, and check if the ATA B is successfully registered with the IPX-2000 as one of the IP extensions.

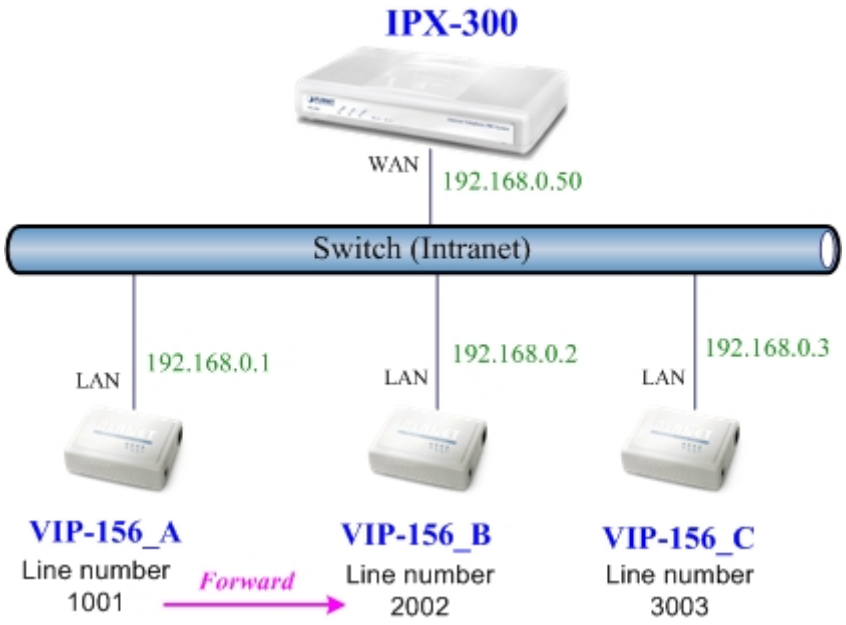
Test the scenario:

1. Pick up the telephone on ATA A
2. Press the keypad: **2002** shall be able to connect to the ATA B
3. Then the telephone set in ATA B should ring. Please repeat the same dialing steps on ATA B to establish the first voice communication from ATA A

Case 5: Call Forward Feature_Example 1

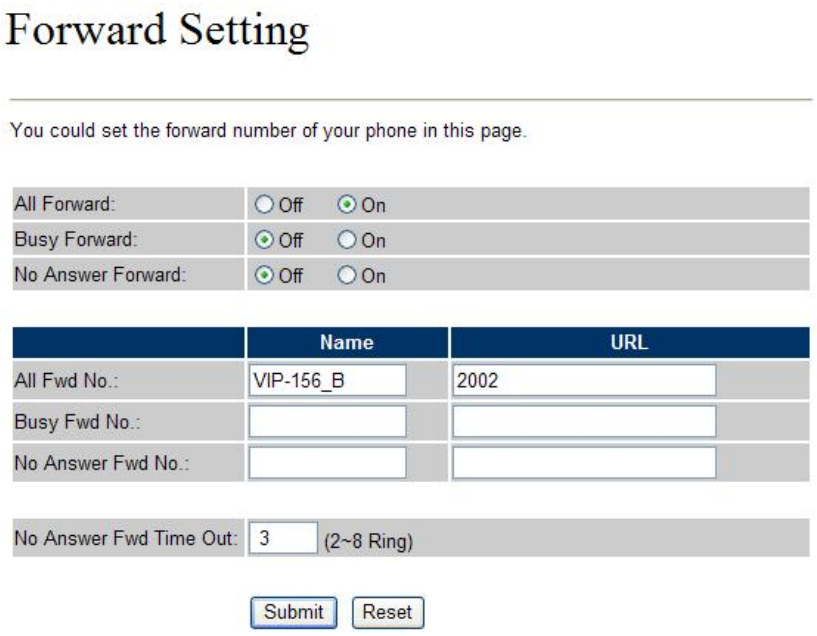
In the following samples, we'll introduce the Call Forward Feature applications.

In this example, there are three VIP-156 register to IPX-300 and VIP-156_A had set Call Forward function to VIP-156_B.



Machine configuration on the VIP-156:

Please log in VIP-156_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the **Name** and **URL** of VIP-156_B, then the sample configuration screen is shown below:

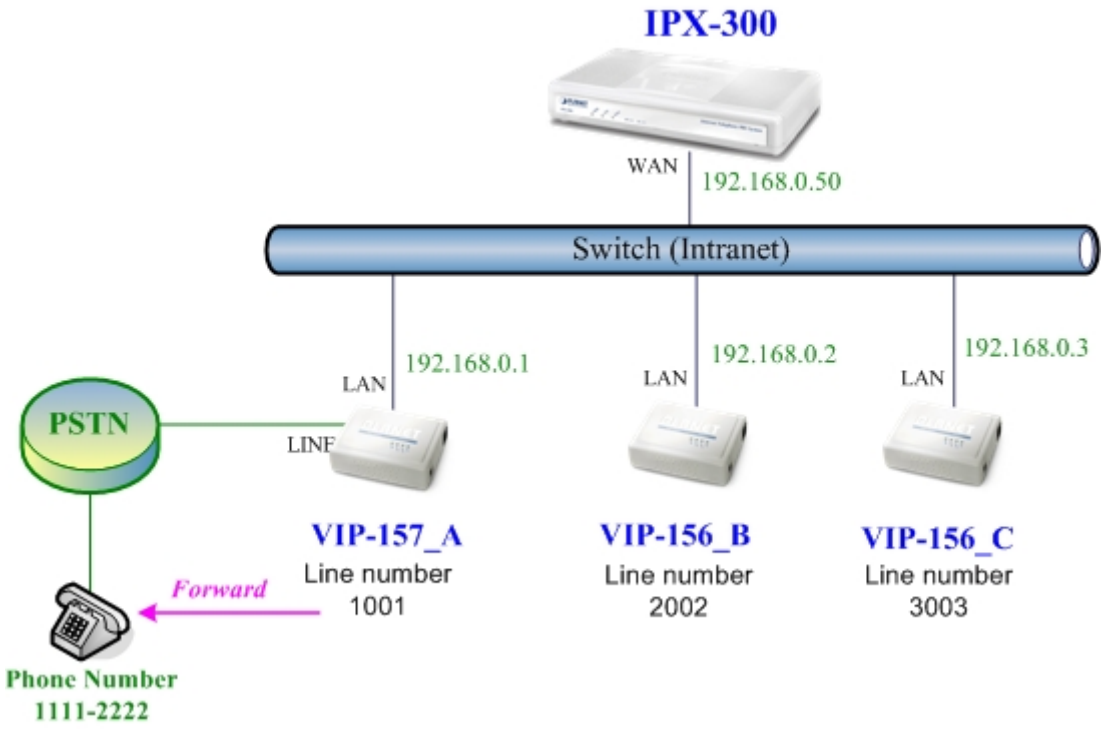


Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the number 1001(VIP-156_A),
3. Because VIP-156_A had set up **All Forward** function to the number 2002(VIP-156_B)
4. The number 2002(VIP-156_B) will ring up then it pick up the telephone and communication with the number 3003(VIP-156_C)

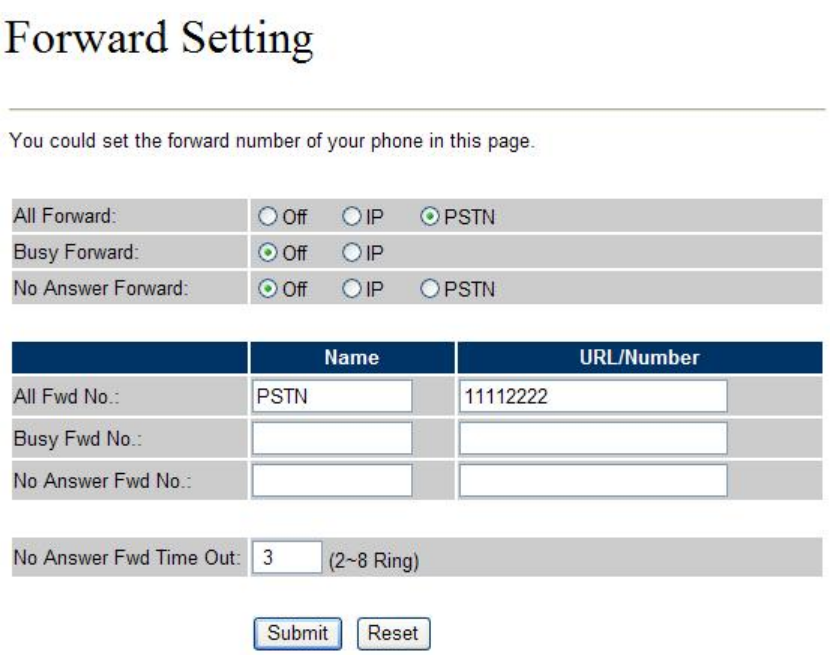
Case 6: Call Forward Feature_Example 2

In this example, there are one VIP-157 and two VIP-156 register to IPX-300. The VIP-157_A had set Call Forward function to phone number 1111-2222 (PSTN).



Machine configuration on the VIP-157:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please select the **All Forward** function to **PSTN** choice and fill in the **Name** and **URL/Number** of PSTN Phone Number 11112222, then the sample configuration screen is shown below:

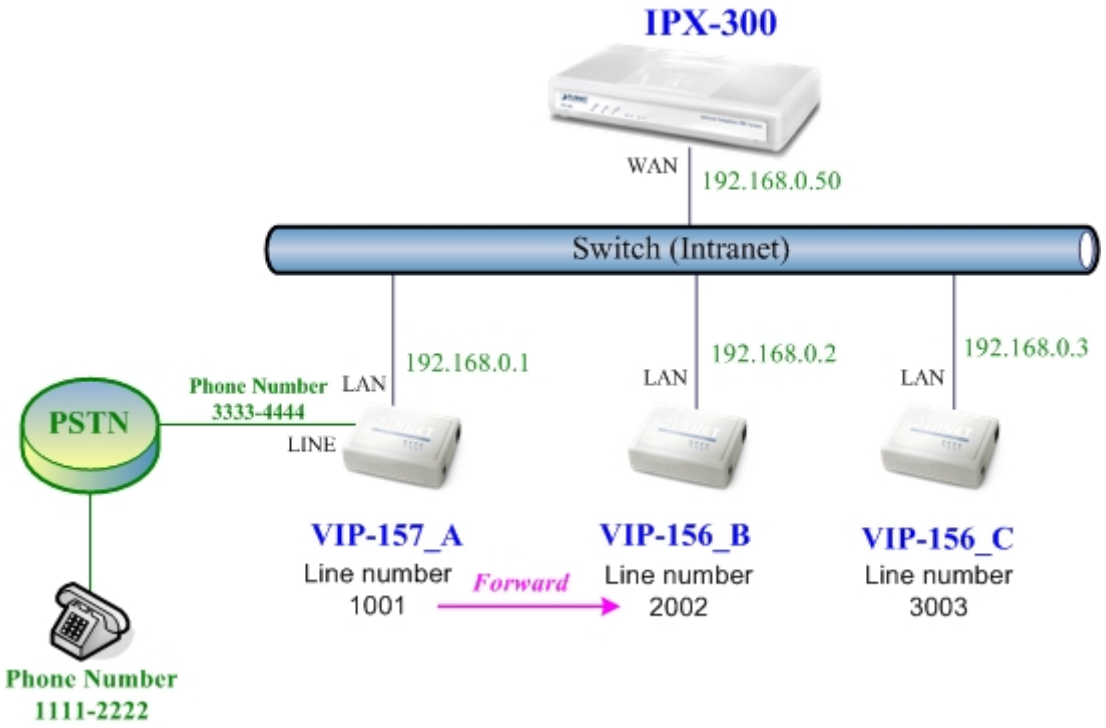

Forward Setting
You could set the forward number of your phone in this page.
All Forward: Off IP PSTN
Busy Forward: Off IP
No Answer Forward: Off IP PSTN
Name
URL/Number
All Fwd No.: PSTN 11112222
Busy Fwd No.:
No Answer Fwd No.:
No Answer Fwd Time Out: 3 (2~8 Ring)
Submit
Reset

Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the number 1001(VIP-157_A)
3. Because VIP-157_A had set up **All Forward** function to the PSTN Phone Number 11112222
4. The PSTN Phone Number 11112222 will ring up then it pick up the telephone and communication with the number 3003(VIP-156_C)

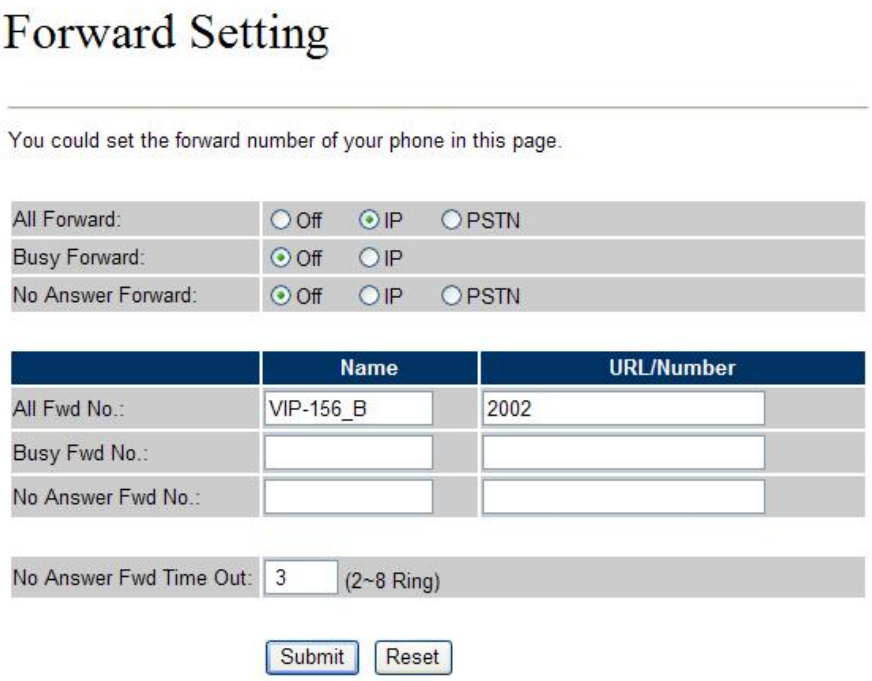
Case 7: Call Forward Feature_Example 3

In this example, there are one VIP-157 and two VIP-156 register to IPX-300. The VIP-157_A had set Call Forward function to number 2002 (VIP-156_B).



Machine configuration on the VIP-157:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please select the **All Forward** function to **IP** choice and fill in the **Name** and **URL/Number** of of VIP-156_B, and then the sample configuration screen is shown below:

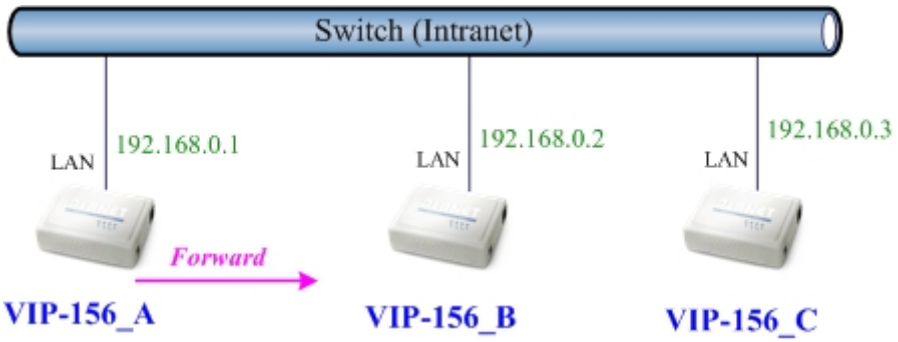

Forward Setting
You could set the forward number of your phone in this page.
All Forward: Off IP PSTN
Busy Forward: Off IP
No Answer Forward: Off IP PSTN
Name
URL/Number
All Fwd No.: VIP-156_B 2002
Busy Fwd No.:
No Answer Fwd No.:
No Answer Fwd Time Out: 3 (2~8 Ring)
Submit
Reset

Test the scenario:

1. PSTN Phone Number 11112222 pick up the telephone
2. Dial the PSTN Phone Number 33334444(VIP-157_A)
3. Because VIP-157_A had set up **All Forward** function to the number 2002(VIP-156_B)
4. The number 2002(VIP-156_B) will ring up then it pick up the telephone and communication with the PSTN Phone Number 11112222

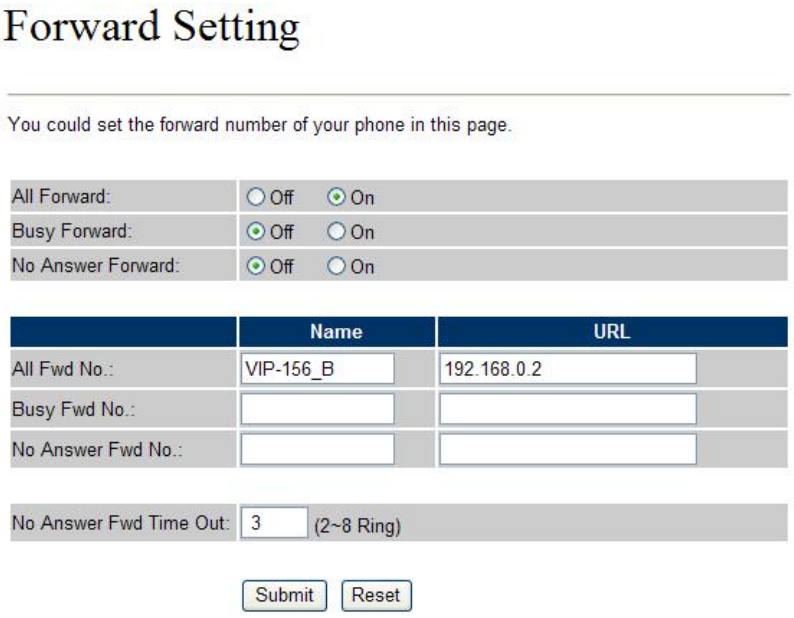
Case 8: Call Forward Feature_Example 4

In this example, there are three VIP-156 and connect with Peer to Peer mode. VIP-156_A had set Call Forward function to VIP-156_B.



Machine configuration on the VIP-156:

Please log in VIP-156_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please enable the **All Forward** function and fill in the **Name** and **URL** of VIP-156_B, and then the sample configuration screen is shown below:

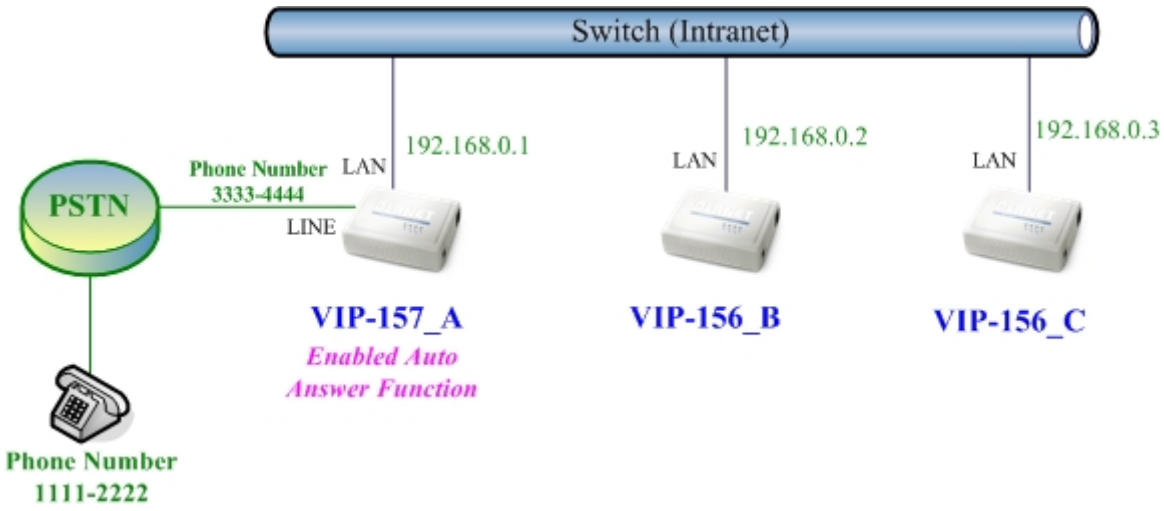


Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the IP Address 192.168.0.1(VIP-156_A)
3. Because VIP-156_A had set up **All Forward** function to the IP Address 192.168.0.2 (VIP-156_B)
4. The IP Address 192.168.0.2 (VIP-156_B) will ring up then it pick up the telephone and communication with the VIP-156_C

Case 9: Auto Answer Feature_IP to PSTN

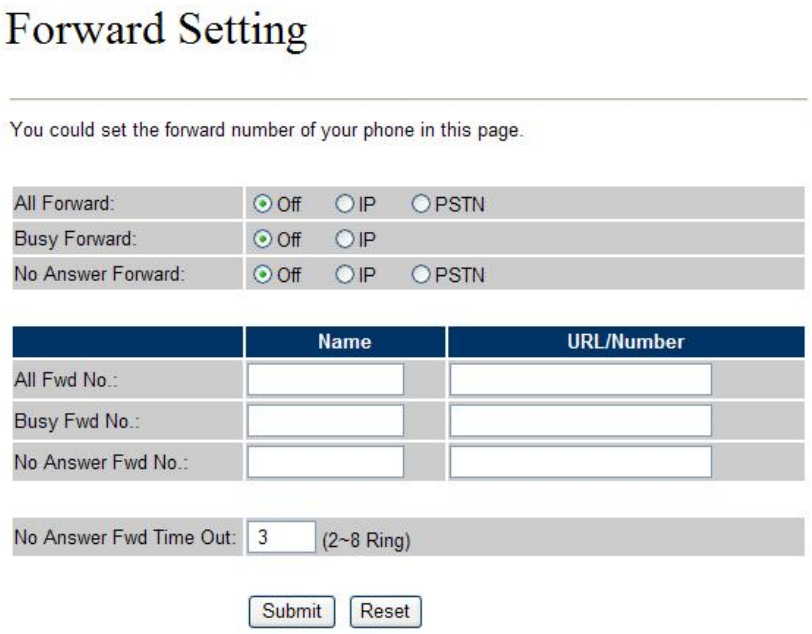
In this example, there are one VIP-157 and two VIP-156 and connect with Peer to Peer mode. The VIP-157_A had set Auto Answer function for forwarding calls to arbitrary telephone. If there have incoming IP calls and VIP-157_A doesn't answer the incoming calls after specific time, the caller will hear prompt sounds to input the password then dial out an arbitrary PSTN telephone.



Machine configuration on the VIP-157:

STEP 1:

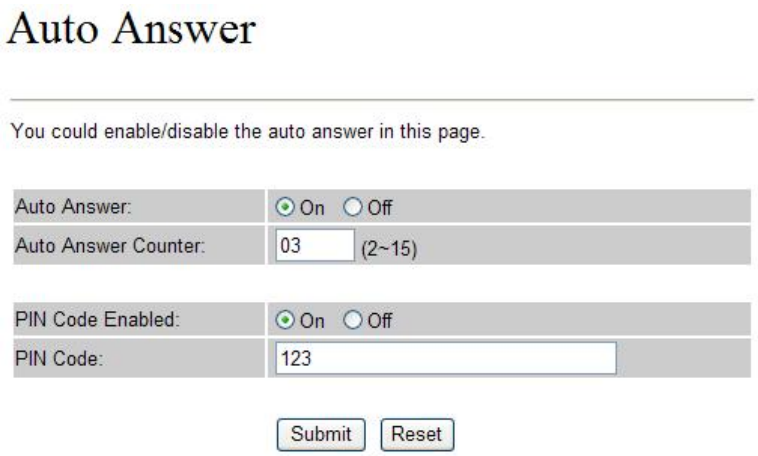
Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Call Forward** config menu. In the setting page, please disable All **Forward** function, and then the sample configuration screen is shown below:



STEP 2:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select

the **Auto Answer** config menu. In the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, then the sample configuration screen is shown below:

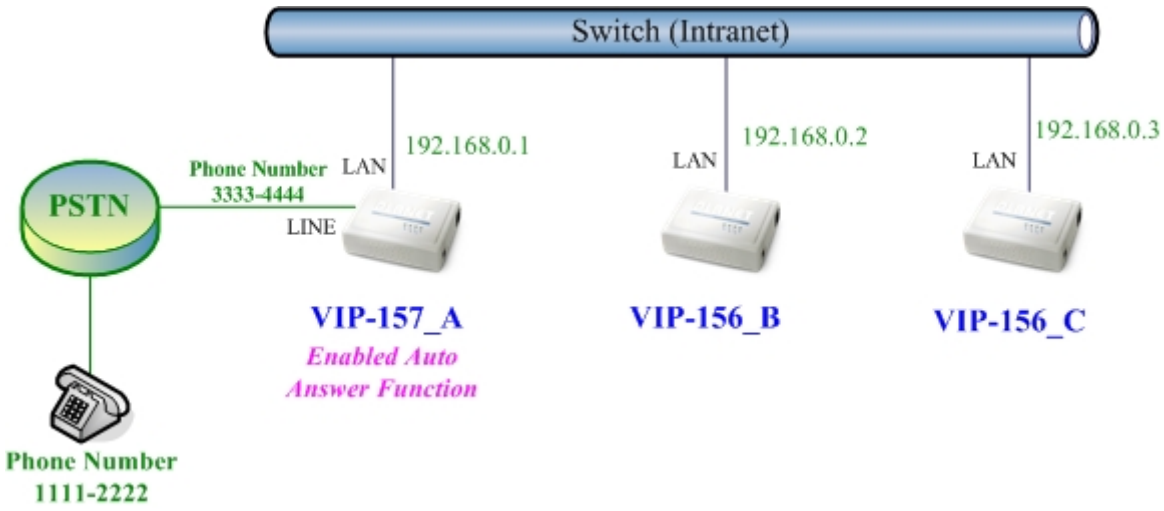


Test the scenario:

1. VIP-156_C pick up the telephone
2. Dial the IP Address 192.168.0.1(VIP-157_A)
3. VIP-157_A will ring up but doesn't answer the call
4. After **3** rings, the VIP-156_C will hear the prompt sounds then input the password **123#**
5. VIP-156_C will hear the dial tone from PSTN line then input Phone Number 11112222
6. The Phone Number 11112222 will ring up then it pick up the telephone and communication with the VIP-156_C

Case 10: Auto Answer Feature_PSTN to IP

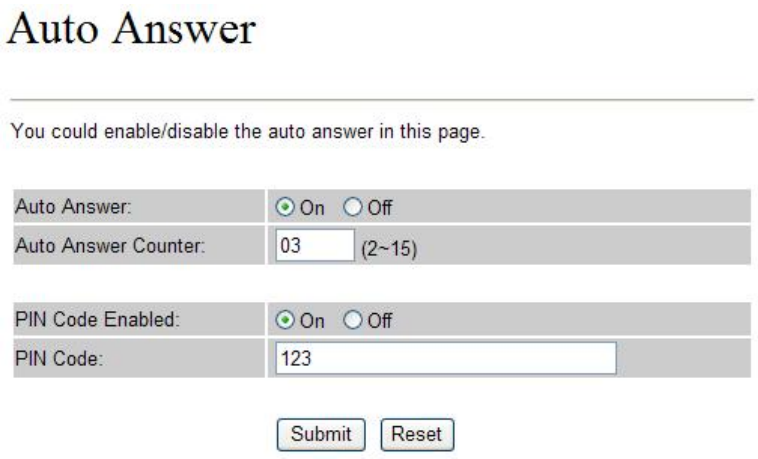
In this example, there are one VIP-157 and two VIP-156 and connect with Peer to Peer mode. The VIP-157_A had set Auto Answer function for forwarding to arbitrary telephone. If there have incoming PSTN calls and VIP-157_A doesn't answer the incoming calls after specific time, the caller will hear prompt sounds to input the password and then dial out an arbitrary IP telephone.



Machine configuration on the VIP-157:

STEP 1:

Please log in VIP-157_A via web browser, browse to the **Phone Settings** menu and select the **Auto Answer** config menu. In the setting page, please enable the **Auto Answer** and **PIN Code Enabled** function, and then the sample configuration screen is shown below:



STEP 2:

Please log in VIP-157_A via web browser, browse to the **Phone Book** menu and select the **Speed Dial Settings** config menu. In the setting page, please add a speed dial number for dial to IP address 192.168.0.2 (VIP-156_B), and then the sample configuration screen is shown below:

Test the scenario:

1. The Phone Number 11112222 pick up the telephone
2. Dial the PSTN Phone Number 33334444(VIP-157_A)
3. VIP-157_A will ring up but doesn't answer the call
4. After **3** rings, the Phone Number 11112222 will hear the prompt sounds then input the password **123#**
5. The Phone Number 11112222 will hear the dial tone then input **0#**
6. The IP address 192.168.0.2 (VIP-156_B) will ring up then it pick up the telephone and communication with the Phone Number 11112222

Appendix B The method of operation guide

In this section, we'll introduce the steps of how to set up some call features of the ATA. Please follow the steps below to utilize those features.

Call Transfer

A. Blind Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out (Press "**transfer**" button) to hold the conversation with B.
3. A press "**#510#**" and hear the dial tone, then input the number of C (Follow by the "**#**" key).
4. C will ring up and A will get the busy tone for prompting to hang up
5. C picks up the handset and conversation with B.

B. Attendant Transfer

1. B call to A and they are in the process of conversation.
2. A carry the transfer function out to hold the conversation with B.
3. A press "**#511#**" and hear the dial tone, then input the number of C (Follow by the "**#**" key).
4. C will ring up.
5. C picks up the handset and conversation with A.
6. A hang up and C conversation with B.

3-Way Conference

1. A and B are in the process of conversation.
2. A want to invite C to join their conversation.
3. A press "**Transfer**" or "**Hold**" button to hold the conversation with B at first, then press "**#512#**" and hear the dial tone, then input the number of C (plus the "**#**" key).
4. C will ring up and pick up the handset to conversation with A.
5. A press "**Transfer**" button again, and they will entry the 3-Way conference mode.

Call Waiting

1. A and B are in the process of conversation.
2. C call to A and A will hear the prompt sounds.
3. A press "**Hold**" button to hold the conversation with B, and switch to conversation with C.

Switch the Realm (Registration Proxy Server)

ATA can register to three different SIP Proxies at the same time. It can receive any one of different SIP accounts incoming call, and it can switch to any one SIP accounts for making calls through input

the switch code.

Realm switch code:

1*: Realm 1

2*: Realm 2

3*: Realm 3

For example: The default is realm 1, input the **2*** (Follow by the # key) from keypad and hang up the telephone set. It will switch to realm 2, and it can make the SIP calls via realm 2.

Auto Update firmware by manual (Keypad)

If pick up the handsetof ATA, it will hear the “DoDoDo” prompt. If want to carry out the upgrade action, please input ”**#190#**” to unlock the device at first. Then input ”**#160#**” to upgrade the new firmware.

Appendix C VIP-156/VIP-156PE/VIP-157/VIP-157S Specifications

Product	SIP Analog Telephone Adapter			
Model	VIP-156	VIP-156PE	VIP-157	VIP-157S
Hardware				
LAN	1 x 10/100Mbps RJ-45 port (802.3af PoE for VIP-156PE)			
PC	1 x 10/100Mbps RJ-45 port			
FXS (for telephone set connection)	1 x RJ-11			2 x RJ-11
FXO (PSTN connection)	---		1 x RJ-11	---
Protocols and Standard				
Standard	SIP 2.0 (RFC3261)			
Voice codec	G.711a/u, G.723.1 (6.3k/5.3k), G.726, G.729A, G.729B, GSM			
Fax support	T.38			
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Acoustic echo canceller (AEC) G.165: Line echo canceller (LEC) Jitter Buffer			
Protocols	SIP 2.0 (RFC-3261), TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE			
Network and Configuration				
Access Mode	Static IP, PPPoE, DHCP			
Management	Web, keypad			
Dimension (W x D x H)	94 x 72 x 30 mm			
Operating Environment	0~40 degree C, 10~95% humidity			
Power Requirement	12V DC			
EMC/EMI	CE, FCC Class B			

EC Declaration of Conformity

For the following equipment:

*Type of Product : SIP Telephone Adapter
*Model Number : VIP-156

* Produced by:
Manufacturer's Name : **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.
For the evaluation regarding the R&TTE, the following standards were applied:

Emission		
Conducted / Radiated	EN 55022	(1998 + A1:2000 Class B)
Harmonic	EN 61000-3-2	(1995 Class A)
Flicker	EN 61000-3-3	(1995)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995)
RS	EN 61000-4-3	(1995)
EFT/ Burst	EN 61000-4-4	(1995)
Surge Test	EN 61000-4-5	(1995)
CS	EN 61000-4-6	(1996)
Magnetic Field	EN 61000-4-8	(1993)
Voltage Disp	EN 61000-4-11	(1994)
Safety	EN 60950 3rd	(2000)

Responsible for marking this declaration if the:

☒ **Manufacturer** ☐ **Authorized representative established within the EU**

Authorized representative established within the EU (if applicable):

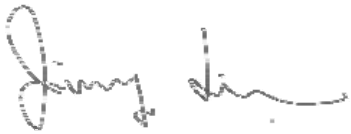
Company Name: **Planet Technology Corp.**

Company Address: **11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

Person responsible for making this declaration

Name, Surname **Jimmy Lin**

Position / Title : **Product Manager**



Taiwan	**7th July, 2005**	
Place	*Date*	*Legal Singnature*

EC Declaration of Conformity

For the following equipment:

*Type of Product : PoE SIP Telephone Adapter
*Model Number : VIP-156PE

* Produced by:
Manufacturer's Name : **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to 1999/5/EC R&TTE.
For the evaluation regarding the R&TTE, the following standards were applied:

Emission		
Conducted / Radiated	EN 55022	(1998 + A1:2000 Class B)
Harmonic	EN 61000-3-2	(1995 Class A)
Flicker	EN 61000-3-3	(1995)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995)
RS	EN 61000-4-3	(1995)
EFT/ Burst	EN 61000-4-4	(1995)
Surge Test	EN 61000-4-5	(1995)
CS	EN 61000-4-6	(1996)
Magnetic Field	EN 61000-4-8	(1993)
Voltage Disp	EN 61000-4-11	(1994)
Safety	EN 60950 3rd	(2000)

Responsible for marking this declaration if the:

☒ **Manufacturer** ☐ **Authorized representative established within the EU**

Authorized representative established within the EU (if applicable):

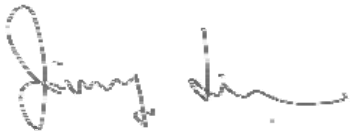
Company Name: **Planet Technology Corp.**

Company Address: **11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

Person responsible for making this declaration

Name, Surname **Jimmy Lin**

Position / Title : **Product Manager**



Taiwan *Place*

7th July, 2005 *Date*

Legal Singnature

EC Declaration of Conformity

For the following equipment:

*Type of Product : VoIP Analog Telephone Adapter (1*FXS + 1*FXO)
*Model Number : VIP-157

* Produced by:
Manufacturer's Name : **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive on (89/336/EEC,92/31/EEC,93/68/EEC).
For the evaluation regarding the EMC, the following standards were applied:

Conducted / Radiated	EN 55022	(1998 + A1:2000 + A2:2003)
Harmonic	EN 61000-3-2	(2000)
Flicker	EN 61000-3-3	(1995 + A1:2001)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995 + A1:1998 + A2:2000)
RS	EN 61000-4-3	(2002 + A1:2002)
EFT/ Burst	EN 61000-4-4	(1995 + A1:2000 + A2:2001)
Surge Test	EN 61000-4-5	(1995 + A1:2000)
CS	EN 61000-4-6	(1996 + A1:2000)
Magnetic Field	EN 61000-4-8	(1993 + A1:2000)
Voltage Disp	EN 61000-4-11	(1994 + A1:2000)

Responsible for marking this declaration if the:

☒ **Manufacturer** ☐ **Authorized representative established within the EU**

Authorized representative established within the EU (if applicable):

Company Name: **Planet Technology Corp.**

Company Address: **11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

Person responsible for making this declaration

Name, Surname **Jimmy Lin**

Position / Title : **Product Manager**

Taiwan	**16 March, 2006**	
Place	*Date*	*Legal Signature*

EC Declaration of Conformity

For the following equipment:

*Type of Product : VoIP Analog Telephone Adapter (2*FXS)
*Model Number : VIP-157S

* Produced by:
Manufacturer's Name : **Planet Technology Corp.**
Manufacturer's Address: 11F, No 96, Min Chuan Road
Hsin Tien, Taipei, Taiwan, R. O.C.

is herewith confirmed to comply with the requirements set out in the Council Directive on the Approximation of the Laws of the Member States relating to Electromagnetic Compatibility Directive on (89/336/EEC,92/31/EEC,93/68/EEC).
For the evaluation regarding the EMC, the following standards were applied:

Conducted / Radiated	EN 55022	(1998 + A1:2000)
Harmonic	EN 61000-3-2	(2000)
Flicker	EN 61000-3-3	(1995 + A1:2001)
Immunity	EN 55024	(1998 + A1:2001)
ESD	EN 61000-4-2	(1995 + A1:2001 + A2:2000)
RS	EN 61000-4-3	(2002 + A1:2002)
EFT/ Burst	EN 61000-4-4	(1995 + A1:2000 + A2:2001)
Surge Test	EN 61000-4-5	(1995 + A1:2000)
CS	EN 61000-4-6	(1996 + A1:2000)
Magnetic Field	EN 61000-4-8	(1993 + A1:2000)
Voltage Disp	EN 61000-4-11	(1994 + A1:2000)

Responsible for marking this declaration if the:

☒ **Manufacturer** ☐ **Authorized representative established within the EU**

Authorized representative established within the EU (if applicable):

Company Name: **Planet Technology Corp.**

Company Address: **11F, No.96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C**

Person responsible for making this declaration

Name, Surname **Jimmy Lin**

Position / Title : **Product Manager**

Taiwan	**17 March, 2006**	
Place	*Date*	*Legal Signature*